A2 Electronics Project: DARPS: A Digital Audio Recorder and Playback System.
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Year: 2011

System Overview:

I will design and create a system that will record a variable amount of audio data and then replay the data through a loud speaker. It will be controlled by two push to make switches; one that will set the device into record mode where the audio will be recorded, and the other which will put the device into playback mode where the recorded audio will be replayed through a loudspeaker. The time will be controllable by a jump lead. If both modes have been enabled no sound should be recorded or played back.

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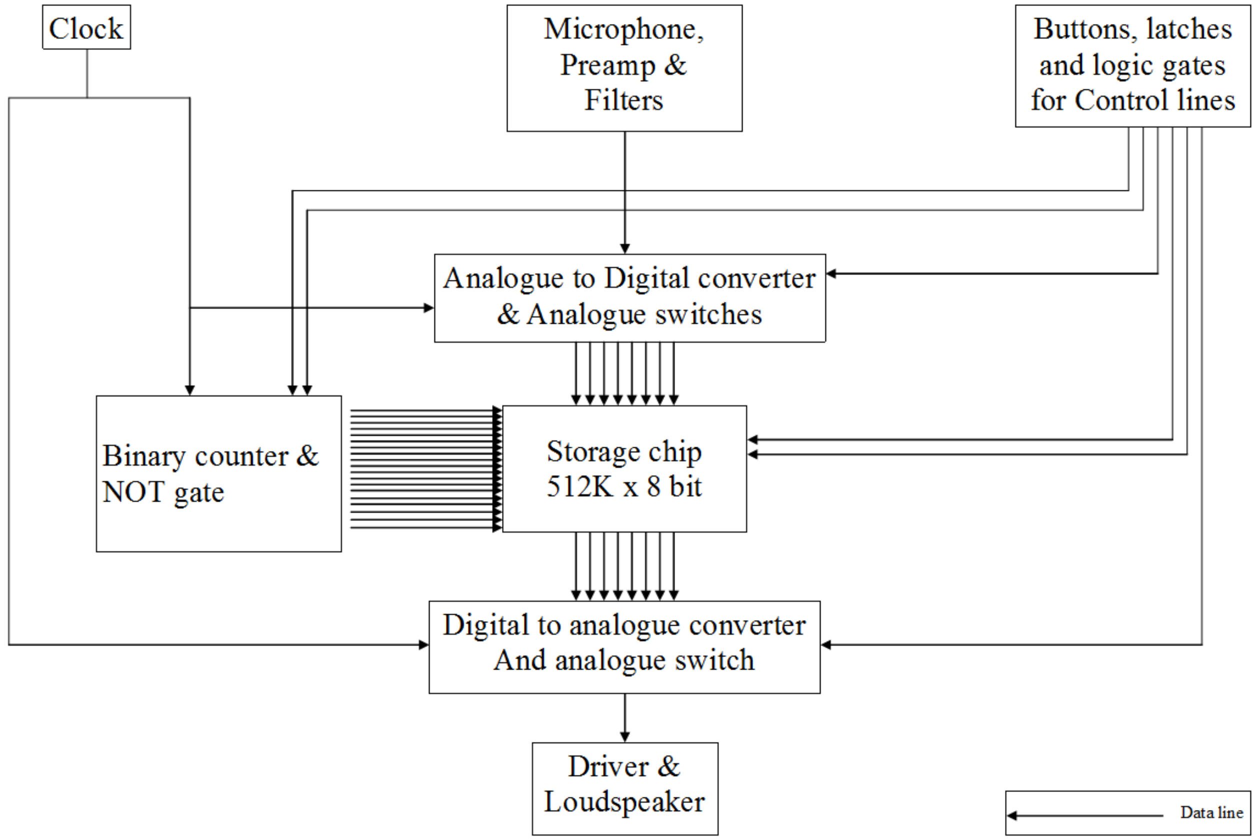
Block Diagram



Detailed Explanation:

The clock is made and set at 8000Hz as this is the frequency I plan to use for this circuit. I will use 8000Hz because with the memory chip that the circuit contains I will be able to record a total of sixty-four seconds of audio data with at this, the main reason it is here is that the conversation time on the ADC is slow and cannot be any faster than 10KHz, so 8 is a safe value. It is also the same quality that is used in phone calls so voices can be clearly heard.

Sound waves are sent into the microphone where it is transferred into electrical energy.

This is then amplified and the high frequencies are cut off at 4000Hz to stop aliasing. This is then sent to the ADC (ADC0804) where it is converted to 8 bit binary data at the clock speed.

The clock is also sent into a binary counter which will be made from two binary counters collected together with the final line on the first counter to go into the clock of the second counter so that I can have enough output lines, which will count up as each clock signal is sent. It will count up until the set amount which will be set by the location of the reset line from

the dual latches. There will be 19 address lines from the counters to the address bus on the chip and the 20th line will go initially to the reset on the latch for full 64 seconds of time. There will be three binary outputs from this, one that controls the Write enable, one that controls the Output enable, and one that controls the analogue switches from the ADC to the memory chip.
When the output is enabled from the memory chip, the 8 bit digital signal is sent into the DAC (ZN425) which is also running off the same clock signal. This should create the analogue wave back in between 0 and 5v as it were before going into the ADC. This then is sent to the driver, which returns the signal to being between -2.5 and 2.5V and then an op amp will amplify it by a quarter so that the input levels will be correct for the secondary amplifier which will amplify it to a maximum of 12V and be able to drive the loud speaker as it can source a lot more current than the primary amplifier or DAC.

Quantitative Specification:

Almost all of the chips will run off a 0v and 5v supply; however there will be a -5 and +12 v supply for two special chips. The -5V supply will be for the two LM741s and the 12V supply for the lm380.
To get a relaxation oscillator at 8000Hz we must use this formula:

$$F = {}^{1}/_{0.5RC}$$

So using a 10kΩ resistor, we need to know what capacitance we want.
So we rearrange it so that

$$C = {}^{1}/_{0.5RF} = {}^{1}/_{(0.5 * 10000 * 8000)} = 25nF$$

Because of a frequency of 8000Hz, the HCF on the preamp must be set to 4000Hz to stop aliasing. The gain on the preamp will be the correct amount to bring the signal to varying between 0V and 5V; with 2.5V being where there is no amplitude.

$$F_{Break} = {}^{1}/_{2\pi RC}$$

Resistor value being 10KΩ, Capacitor value being 3.97nF so

$$F_{Break} = {}^{1}/_{2\pi RC} = {}^{1}/_{(2\pi * 10000 * 3.97 * 10^{-9})} = 4008.94$$

The binary counter will count up to 524287 before resetting to 0; this is the highest number possible as the memory chip has 19 address lines and 2^{19} = 524288. This then will allow approximately 64 seconds of sound to be recorded. However this can be shortened so the user can choose between 64, 32, 16, 8, 4 and 2 seconds by moving the jump lead to the appropriate connection.

The preamp needs to bring the signal from a peak of 0.1v to 2.5v, so the maximum gain we want is 25. For a safe value I chose 22 but as it is an inverting amplifier this is a negative. So to get -22 we must use

$$G = -\frac{R_F}{R_{In}} = -\frac{220000}{10000} = -22$$

After the amplifier there needs to be a voltage offset of 3.2V, this is worked out by wanting to have a voltage offset of 2.5v, but after this it goes through a diode which drops 0.7v, so we need to add this onto the offset. This means that we need to make a potential divider which has 3.2v in the middle.

To do this we use ohms law to work out the current across the entire potential divider if we use 100k and 56k resistors. This comes out to be 32µA, which then across the bottom resistor gives 3.205V for an offset. However, when this is connected you make a subtle low cut filter. So we need to make sure the capacitor is a value which does not interfere with this circuit. For this we also use

$$F_{break} = {}^{1}\!/_{(2\pi * R * C)}$$

We will use 15Hz as an acceptable value for the frequency to be cut at as this is below what we can hear.

Rearranging the equation:

$$F_{break} = {}^{1}\!/_{(2\pi * R * C)}$$

$$C = {}^{1}\!/_{2\pi RF} = {}^{1}\!/_{(2\pi * 100000 * 15)} = 1.06 * 10^{-7} \approx 100nF$$

Around 1.06E-7. This is approximately 100nF.

For the hold down resistors I use 10K because this is a value which will not draw too much current through the circuit. If it was a lower resistance the power would be higher and the circuit less efficient.

My data will be 8bit for many reasons; firstly 8 bit gives me 256 voltage levels, which when between 0 and 5v means that the resolution is 0.02V per division. Secondly the chips all support 8 bit and if it were any higher than 8 bit I would have even more issues with keeping the wiring neat.

For using LEDS when testing I will use a 470 Ohm resistor for each LED so that this may limit the current sufficiently enough to make sure that a led will not be damaged with 5v. V=IR,$\frac{5}{470} = 10\text{mA}$ - perfect for an LED.

For the LEDs actually in the circuit I will use 270 Ohms because this allows 20mA ($\frac{5}{270} = 0.0185$) which allows the LEDs to glow without being damaged but brighter.

The values for the self-clocking of the ADC were given to me by its datasheet.

For the voltage offset which will bring the signal back to varying around 0v, we use a basic RC circuit; however we need to make sure that this circuit does not also turn into a low cut filter that would affect the system. To do this we use a high resistance resistor (330k) and then the formula

$$1/(2\pi * R * C) = f_{break}$$

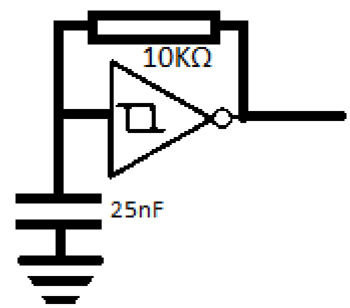
Using a 470nF Capacitor it cuts the frequency at a very safe 1.026Hz
For the first op-amp in the driver; I need a gain of -¼. To work this out we use $G = -\frac{R_f}{R_{in}}$. This is $-27k/_{100k} = -0.27$. This is close enough to ¼ and will work well. It will need a +5 and -5 power supply.
The second Amplifier and loud speaker will need a 12V and 0V power supply, and the gain is controlled by the potentiometer.

Research and Circuit Diagrams:

Subsystem 1 (8KHz Relaxation Oscillator):

$$F = \frac{1}{0.5RC} = \frac{1}{0.5 * 10000 * 25 * 10 - 9} = 8000Hz.$$

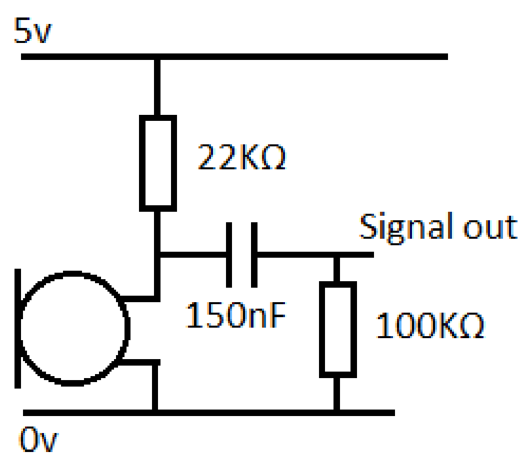
These values would produce a nice frequency for the clock. The clock does not need an enable input as the control of the device will be handled a different method instead of clock control.



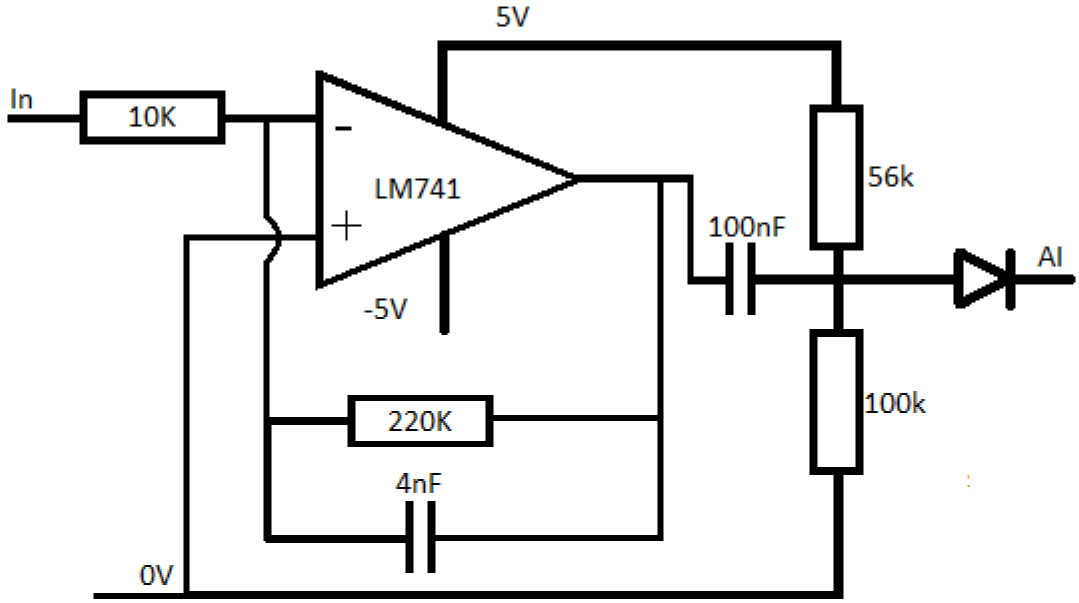
Subsystem 2 (Microphone and Preamp):

Microphone:

The primary resistor has a value of 22K due to testing of the output signal when set at this resistance the best signal was shown, even though the specifications for the microphone say a much lower value, we found at this value the signal was near impossible to detect. We can see with this setup that the sound output being around a maximum of 100mv, this value I will use for the maximum specifications to design my preamp around. The output has an impedance of 100KΩ.
I got the circuit for this from [1]



Preamp:



The input to the preamp will be between -0.1v and 0.1v and for our system we want to have a signal between 0 and 5 volts, so we need to amplify the voltage by a maximum of 25 and then offset the voltage.
Therefore I am using an inverting amplifier with a Bass pass filter, which means I can cut frequencies off at 4000Hz to stop aliasing. Inverting the signal will have no issues as you will not be able to hear the difference and I can re invert it on the driver if necessary.
For this I'm using a low pass filter with the break frequency of 4KHz, to work out the values we use:

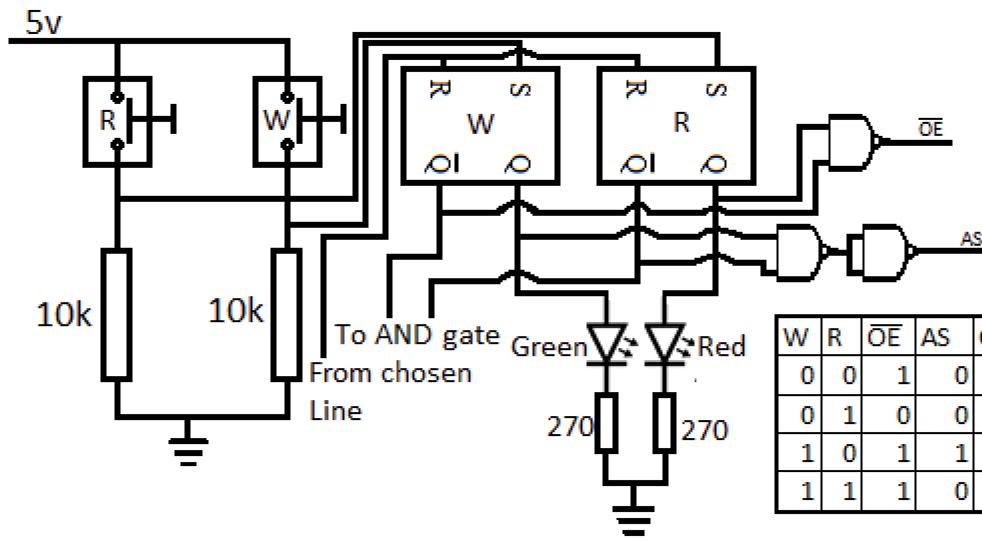
$$F_{break} = {}^{1}/_{2\pi RC}$$

With R being the resistor value and C being the capacitance of the capacitor.
I want the break frequency to be 4000Hz so I set R to be 10KΩand C to be 3.97nF so I will pick a value near that.
Supplying the op-amp with -5 and 5V means that it can amplify up to around -3.5 and 3.5V due to the chip saturating at this point, this is good as the max I want is 2.5V so this will do me perfectly.
As my signal into the preamp peaks at around 0.1V when very loud, and for the ADC I need a voltage between 0 and 5v, I decided that I need to amplify my signal by a maximum of 25; I chose 22 to add a bit of a safety margin to make sure that the operational amplifier does not saturate.
Then it will be offset by 3.2 volts this is because of I need the line to go between 0 and 5v, instead of currently -2.5 and 2.5 volts. However, it is 3.2 because after the offset there is a diode to prevent the voltage from going negative as this can damage the ADC. So while I want an offset of 2.5v, I use 3.2v so that after the diode it will be 2.5v.

$$\frac{V}{R} = \frac{5}{156000} = 32\mu A \ Across\ the\ Potential\ divider.$$
$$IR = V = 32\mu A * 100000 = 3.205V\ Offset.$$

Subsystem 3 (Control lines):

Control lines:



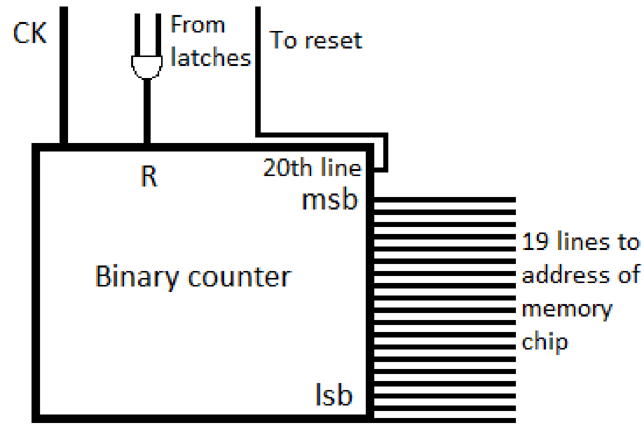
W	R	$\overline{OE}$	AS	GLED	RLED
0	0	1	0	0	0
0	1	0	0	1	0
1	0	1	1	0	1
1	1	1	0	1	1

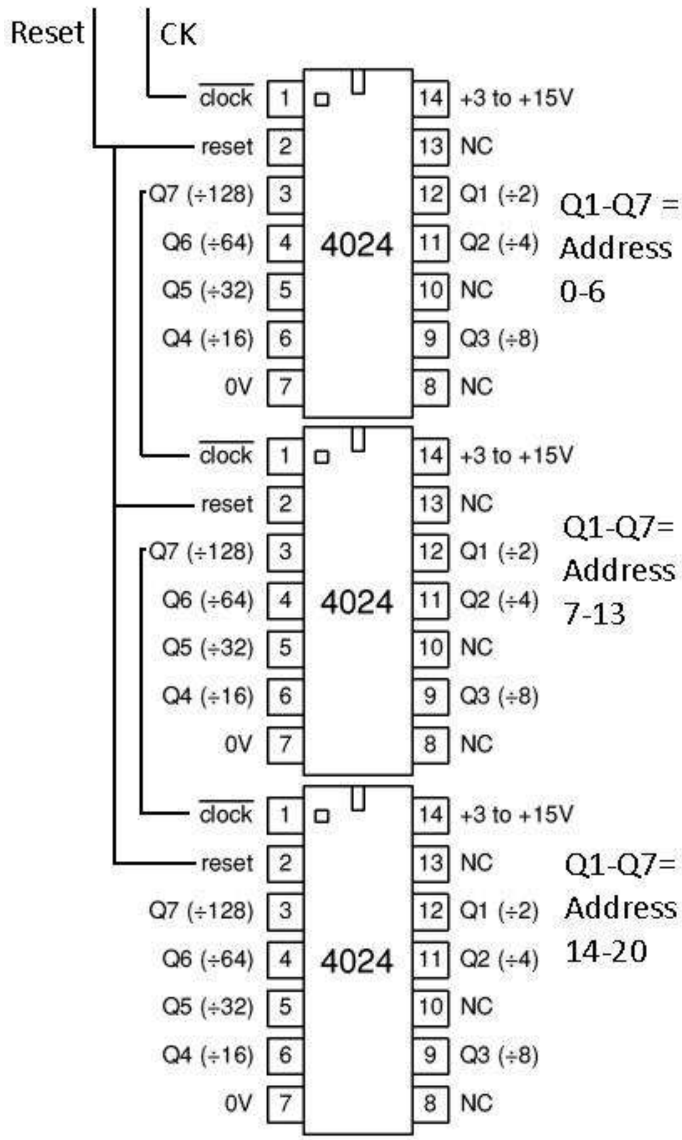
Hold down resistors with Push to make switches, so when pressed each line goes high, when untouched the line remains low. Hold down resistors will be 10kΩ.
A high signal from either of the switches will set one of the flip-flops high, which will enable Q to go high and Q# to go low. There is a NOR gate coming from the Q# outputs so that when either of the lines is low the reset goes low, allowing the counter to count up when connected. It will count up even if both buttons have been pressed because otherwise the system will get stuck, but the OE and WE lines connected the chip are both high so nothing will happen and the Analogue switches will not be enabled.
The output enable line will go low when the Read switch has been pressed but not the Write enable, and the Write enable line will go low when the write line has been pressed but not the read line.
There are two LEDs to show which lines have been pressed and if LEDs are on then the system will not work until bother have gone off, which is after the counter has finished and reset. The colours will be Red for recording and green for playback.

Subsystem 4 (21 Bit Binary Counter):

The counter counts up on falling edge of the clock pulse, and then counts up until it reaches its maximum. Then as the maximum line falls low it triggers a clock pulse on the next counter and this repeats until the address is high enough to enable the reset on the main system to be activated. Having the first input active low is important because it means once the counter has reached the top of its counts, as the MSB goes low, the clock on the next counter goes up.

The reset line goes to the latches from the chosen pin on the counters. If full 64 seconds is wanted, then you chose the 19th address line, 32 seconds the 18th, etc.

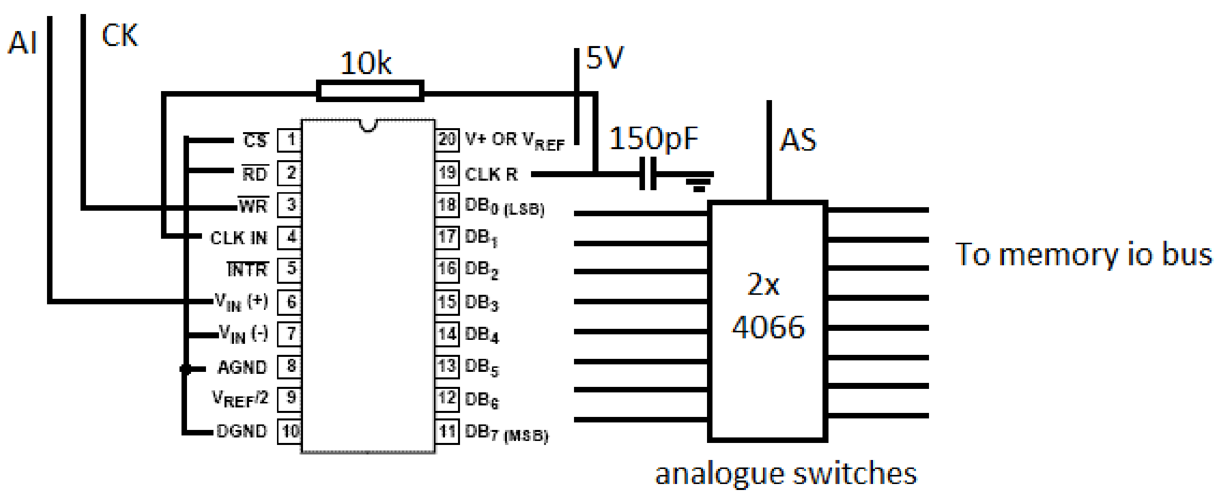




Subsystem 5 (8 Bit Analogue --> Digital Converter and sub-control line):

ADC:

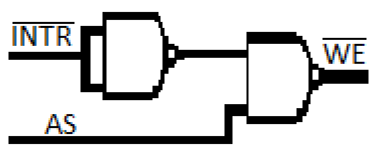
AI is the analogue input from the preamp, it goes through a diode to make sure there are no negative voltages then goes into V_{in}, CS and RD are held low so the chip is enabled. WR# is the clock input so then it will be sent to the output. The data outputs are sent to the analogue switches which are controlled by Q on the binary counters flip flop.
Because WR# is also active low it syncs with the counter instead of being 180 degrees out of phase.



I got the values and design for the self-clocking mechanism

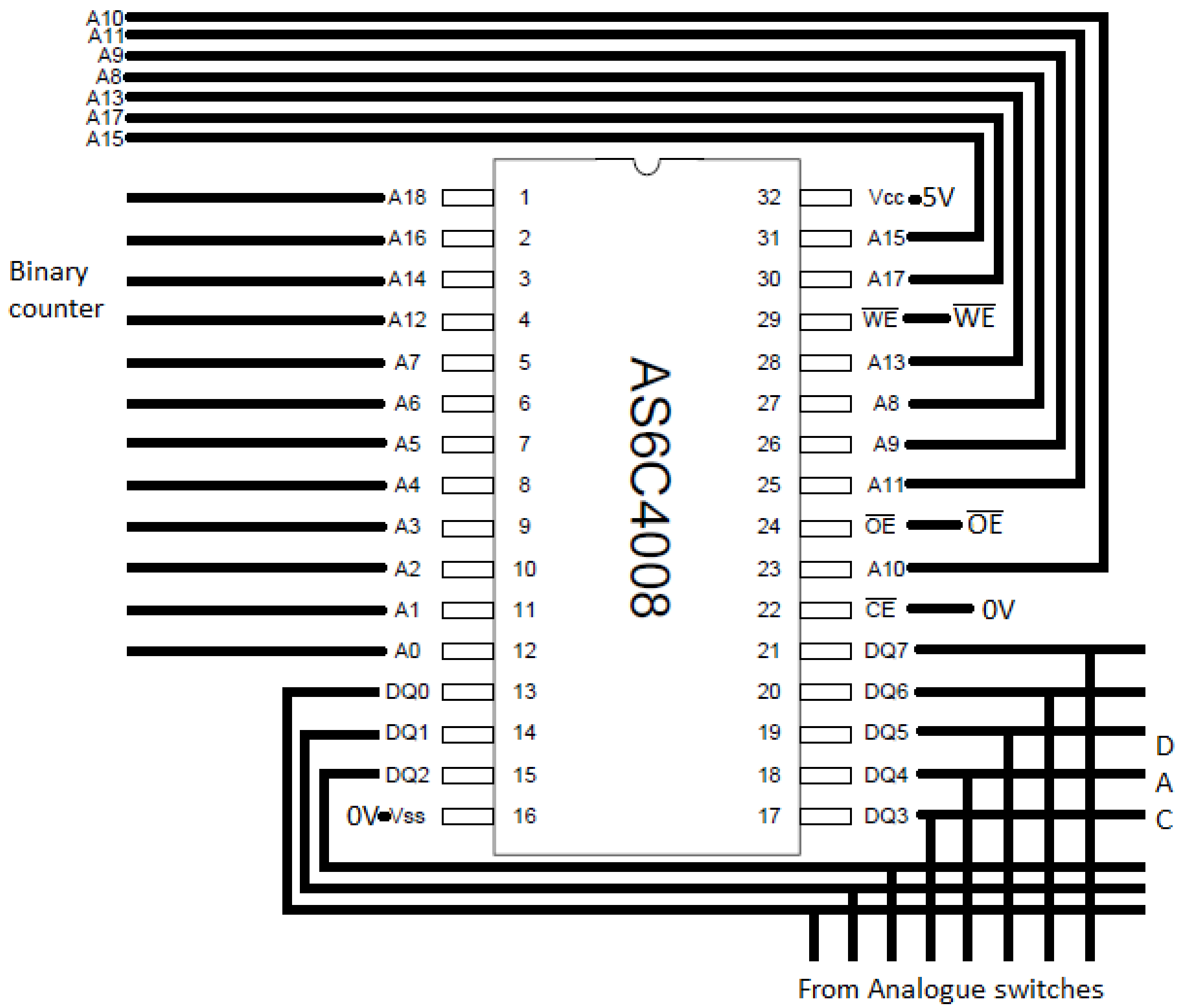
Sub-control line (WE line):

The WE line will be a combination of the write line only being enabled and the conversion from the ADC being ready. It will be done using this logic.



This is because I need an OR gate from the INTR# and line before AS was inverted. But as there would be double inversion I can just remove this line completely and take the AS line. This means that the data is only written to the chip for the moment the conversion is complete, not for the entire time it is recording.

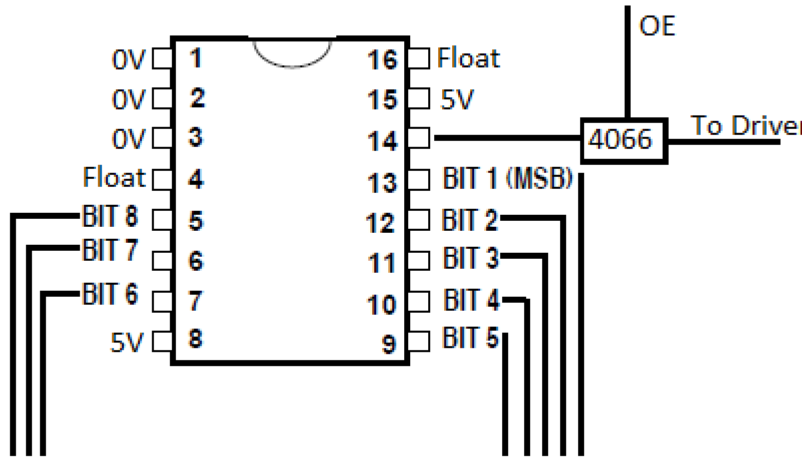
Subsystem 6 (512Kbit*8bit SRAM Volatile memory):



The address lines come into the Chip from the binary counter which makes the addresses that the samples are stored into count up. While the data is inputted on the IO port when the Analogue switches are enabled, but when disabled that line is floating so the output data is not interrupted. The Output enable and Write Enable are active low and controlled from the logic gates in the Binary counter subsystem. CE is always low as chip does not have to be disabled at any time.

As the chip is SRAM, the important part being RA - Random Access. This means that it does not matter which order the address lines go into the chip, as long as it is replayed in the same order, it will still count up and down through the samples. This means that wiring will be a lot easier as each wire just has to go to any address, not the specific one.

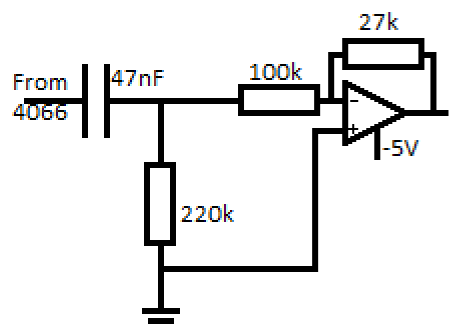
Subsystem 7 (8 Bit Digital --> Analogue converter):



The digital lines will go into the inputs on the DAC.
As the chip is a DAC and an ADC then you must set pin two low to make sure that it is set to DAC mode. This will allow a single analogue out to go into an analogue switch so that while recording you cannot hear playback of what you are recording.
The bit arrangement must line up with the bits on the ADC but it does not matter which location they come from on the memory chip, as long as it is the same line from the ADC.

Subsystem 8 (Driver and loudspeaker):

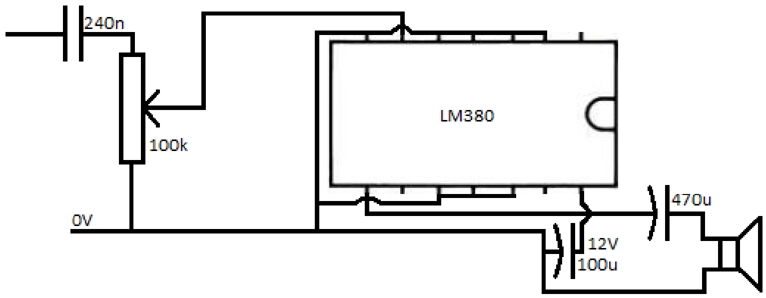
Driver part one:



From my DAC I will get waves that go in between 0V and 5V. First of all I need to take my signal and bring it back down to having the middle voltage as 0v. I will do this by using a coupling capacitor connected to the ground. Making sure I chose correct values to make sure that I do not make this a low cut filter that would affect the signal.
Then I'll need to amplify this voltage and have a power amplifier so it may be able to replay through the loud speaker. To do this I use two OP amp chips. The first one is to allow the second one to be functional.
Firstly I use a 741 with a gain of ¼ to bring the signal level down to between -0.5 and 0.5V, as this is the signal that is needed for the second amplifier circuit.

Driver part two and loudspeaker:

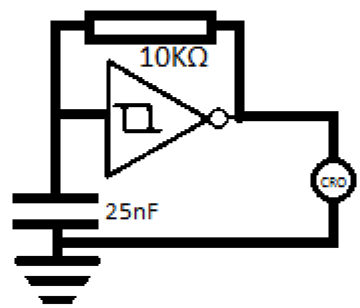
Connects to the output of the driver and connect in between it and the ground. It will convert the electronic signals into audio.



The signal goes in through a coupling capacitor and then through a 100k multi-turn potentiometer, this is effectively the volume control or gain control. When this is at max the sound should be loudest and when at minimum there should be no sound from the speaker. The chip has its own inbuilt feedback loops near the gain of 50. The minimum power to this chip is 12v so this chip will need a separate power rail.
I found the schematic for this part of my system in [1]

Testing Regime:

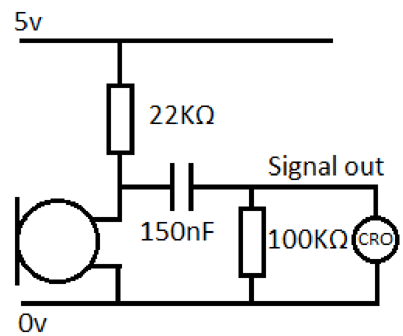
Subsystem 1 (8KHz Relaxation Oscillator):



I will connect the output to an oscilloscope and check the frequency that it is outputting and check that it is a digital signal that would not damage any components (0v and 5v)

Subsystem 2 (Microphone and Preamp):

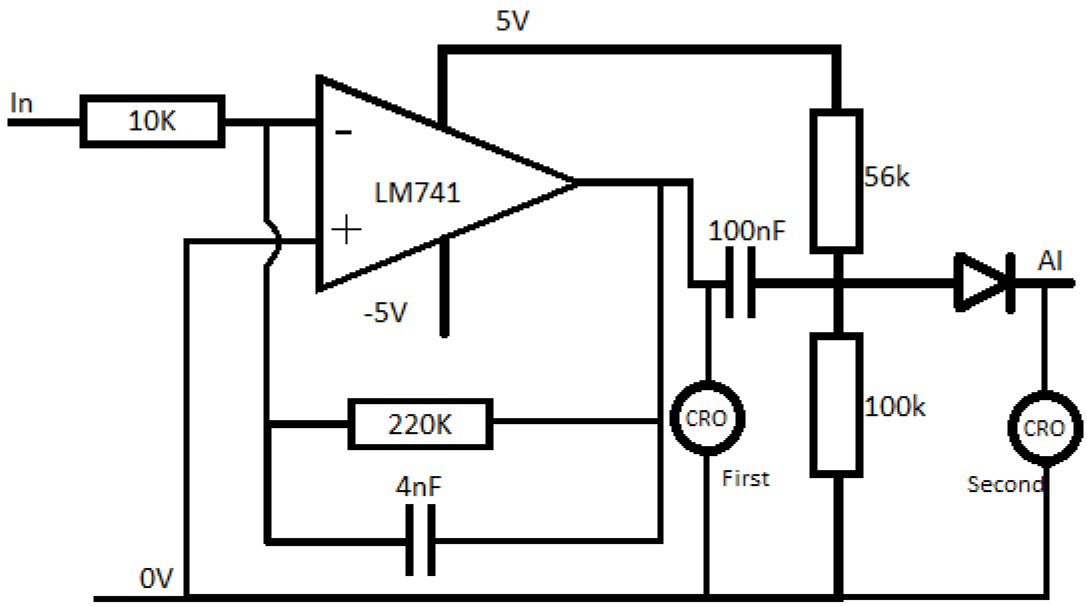
Microphone:



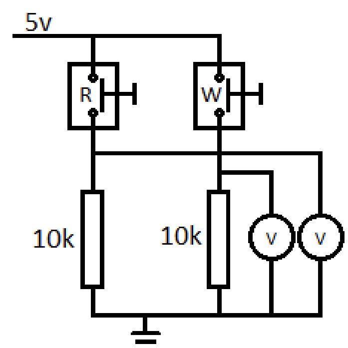
I will connect an oscilloscope to the output and see when making noise with it if the signal goes between -0.1 and 0.1 volts. The oscilloscope will be connected as shown

Preamp:

I will connect a sine wave generator to the input with a low input and an oscilloscope to the output to check that the gain and then test that the voltage offset is the correct amount, I will also turn up the frequency to 4000Hz, 5000Hz and 6000Hz to check that the filter does indeed cut off the high frequencies. Then I will connect them microphone to this and make noise into it to see if the levels are correct and there is no saturation.

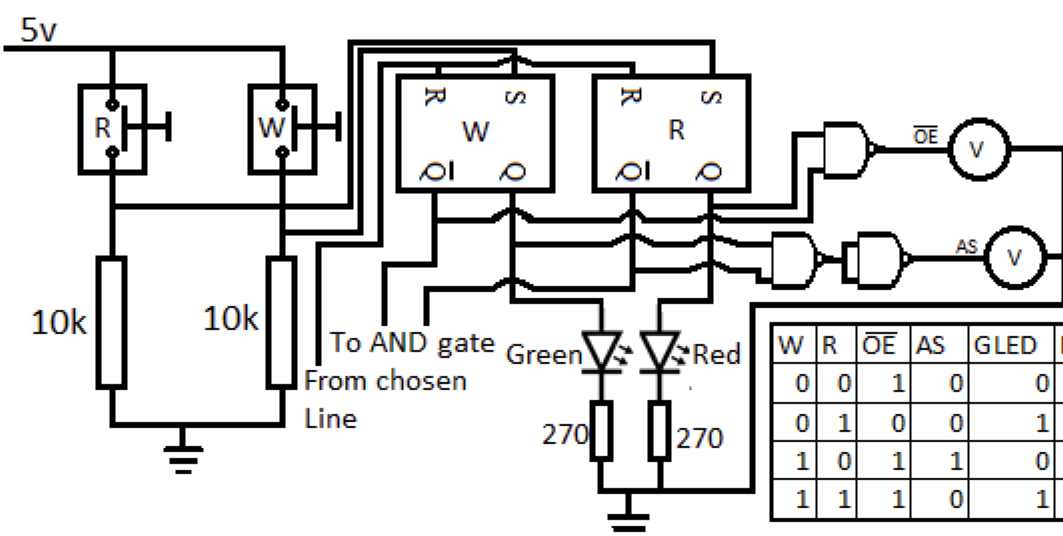


Subsystem 3 (Control lines):



I will put a voltmeter at the output of each line from switches to check that it is low when not pressed and high when pressed.

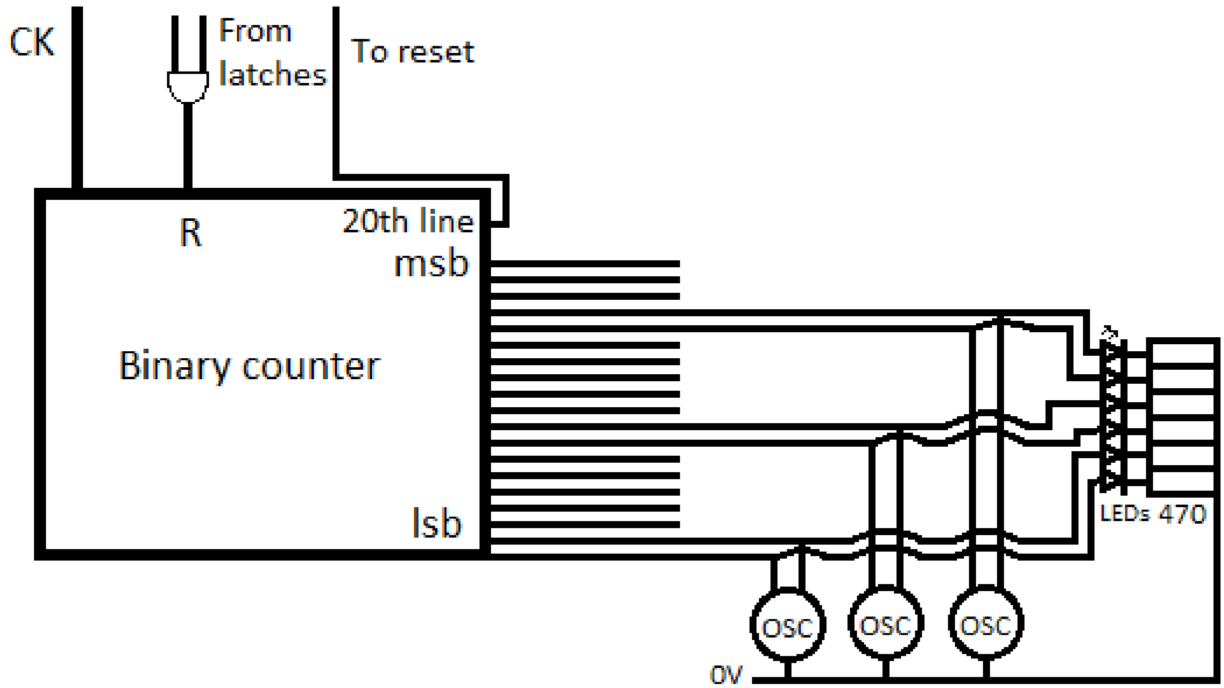
Then I will connect the two latches and LEDs output, and connect these to the control lines and check that they go high, and then can be reset by putting the reset line high.
Then I will connect the NAND gates and check that the outputs of them are correct with a multi-meter when the correct latches are set high and low.



W	R	OE	AS	GLED	RLED
0	0	1	0	0	0
0	1	0	0	1	0
1	0	1	1	0	1
1	1	1	0	1	1

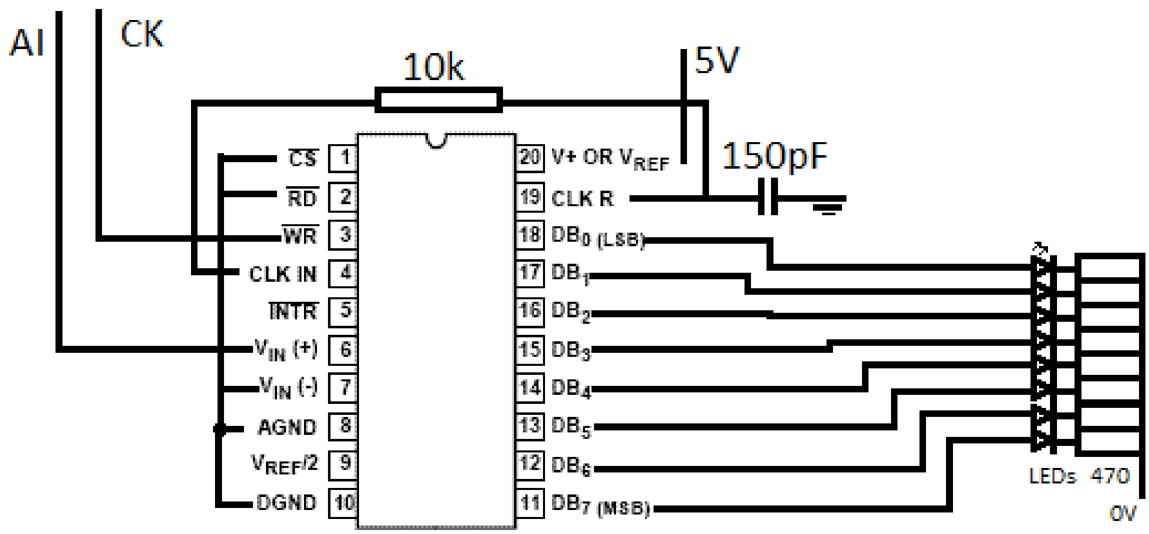
Subsystem 4 (21 Bit Binary Counter):

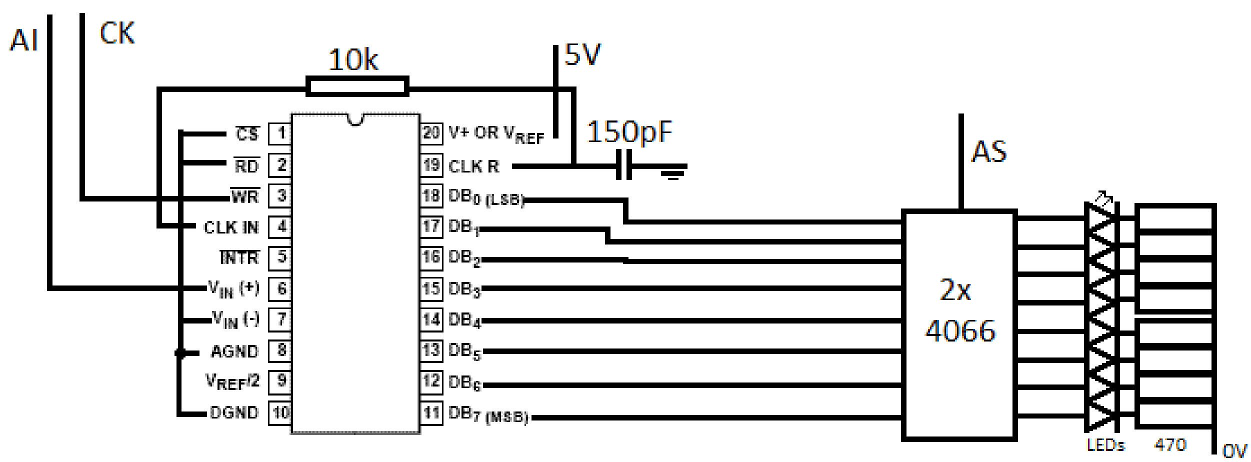
Then I will create the binary counter from the three binary counters, with the clock input set up, so that I have at least 20 data lines, and connect a few of them up to the input of an oscilloscope and check the timing, as well as connecting the 14th line to the reset to the latches and count for a second for the loop to complete. Then placing it on the 20th and checking that it takes 64 seconds as well as repeating this with all the possible timings down to 2 seconds.



Subsystem 5 (8 Bit Analogue --> Digital converter):

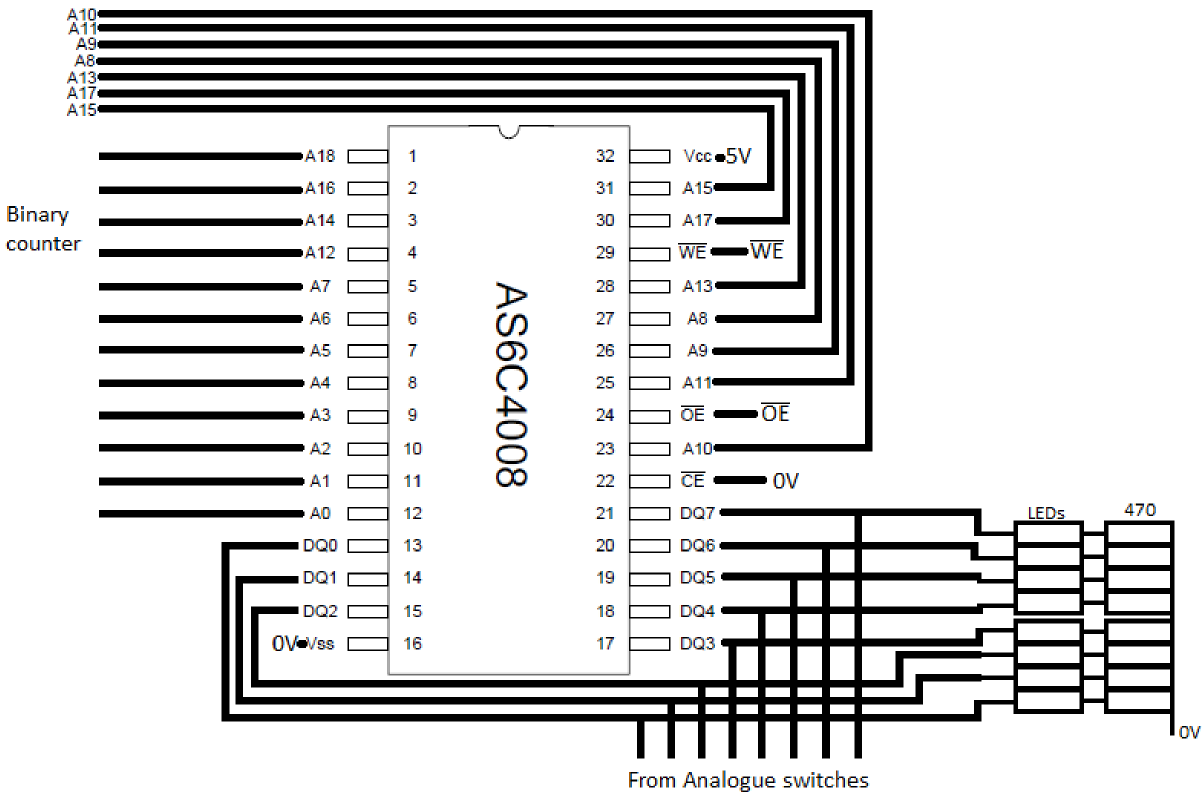
First I will connect up the ADC and put a bar graph displays with resistors attached to the output. I will connect the clock input in and connect the analogue signal in, then making a noise I will hopefully see the signal go up and down on the LEDs in the bar-graph display.
I will then connect the analogue switches and move the bar graph LEDs to the other side of the switches and connect it up to the enable line, and test that the signal still goes though when enabled and is floating when not enabled.



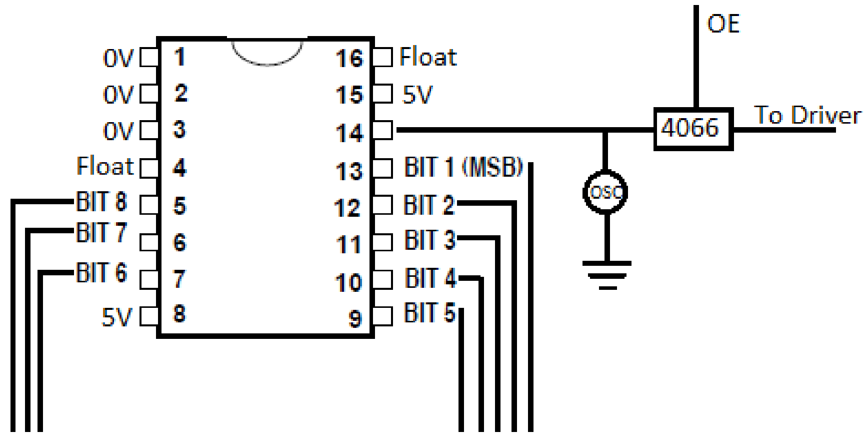


Subsystem 6 (512Kbit*8bit SRAM Volatile memory):

Firstly I will connect the WE# line with the NAND gates and check that I get the right output from both of them being high/low so that write enable is only enabled at the correct time. Then to test the memory chip itself, Then I will connect the output of the analogue switches to it, and tap into the microphone a pattern. This then will show up on some LEDs connected to the output, in the same pattern and timing as I put it in.



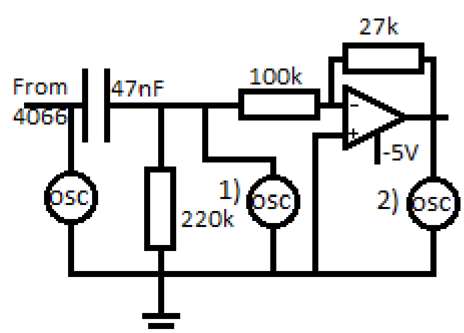
Subsystem 7 (8 Bit Digital --> Analogue converter):



I will connect up the DAC chip and then connect 8 jump leads to the inputs; first with them all grounded and then change each one individually with a volt meter on the output check to the voltage is correct for the binary input. Then I will connect it to the memory chip and check that it outputs the stored pattern correctly.
Then I will connect up an oscilloscope on the output and the input to the ADC and see how different the signal is between the two.

Subsystem 8 (Driver and Loudspeaker):

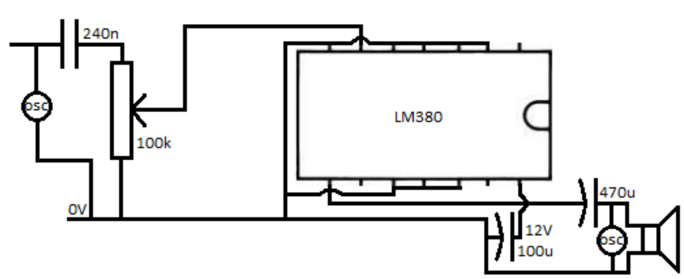
Driver part one:



Firstly I will test that the voltage is brought down from in-between 0v and 5v to -2.5v and 2.5v. To do this I will connect up an oscilloscope both sides of the RC circuit. Then I will connect the signal into the op-amp and out the other side and check that the gain and signal levels are the correct amount.

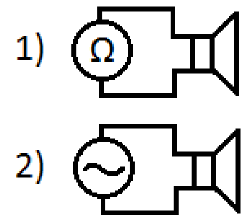
I will do this my connecting an oscilloscope to the input and output of the op amp.

Driver part two and loudspeaker:



Then with this signal I will put it through the potentiometer and into the second amplifier, and on the output I will put the loudspeaker. I will input a small sine wave into the amplifier with the same output levels as the first amplifier from a signal generator and see if the sound is displayed.

To test the loudspeaker I will first check that the value of impedance I have for it is correct by connecting it to an ohm meter. Then I will connect it to a signal generator to hear if it makes a noise and the pitch changes as the frequency does.



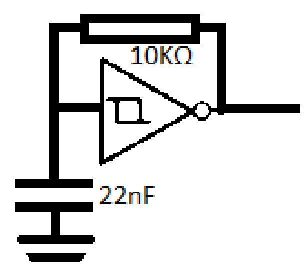
Entire System Testing:

Then I will connect a sine wave from a signal generator into the input of the ADC, and record this into the system, changing the frequencies, and then I will replay this sound and listen to the output of the speaker. If this is successful I will speak into the microphone and then listen to it playing back. I will also connect an oscilloscope on the sine wave to see difference between the input and output wave.

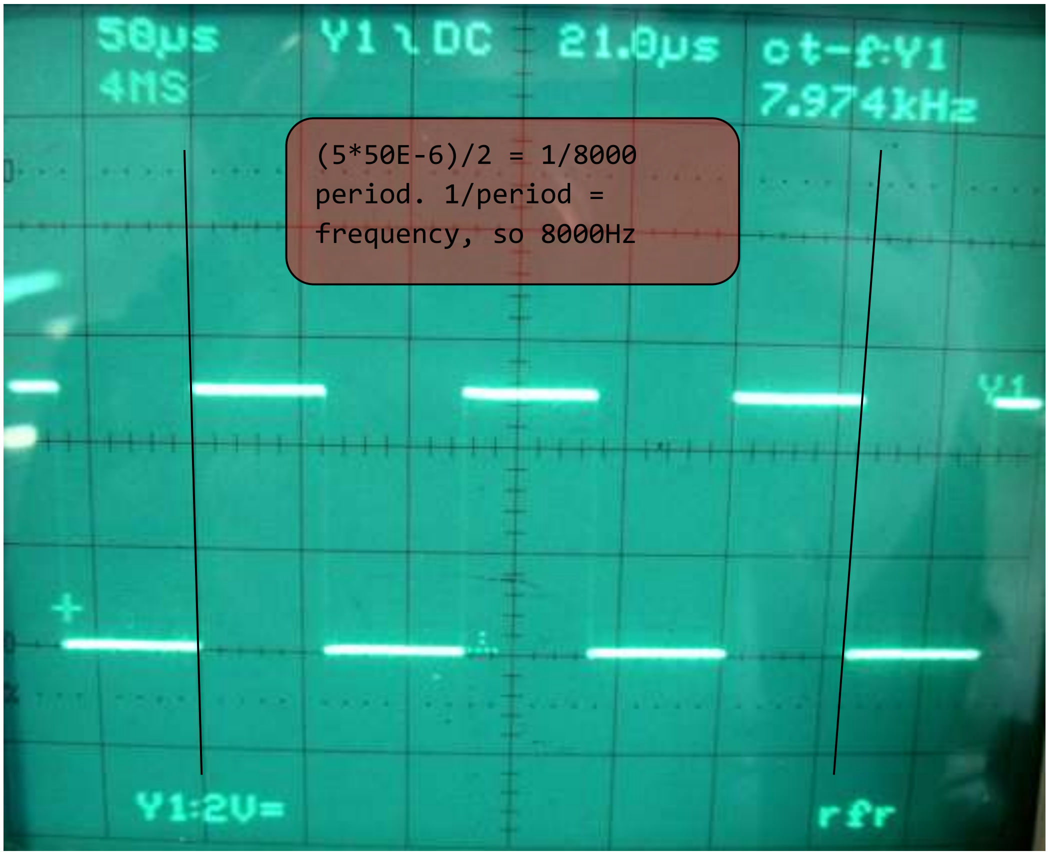
System Testing:

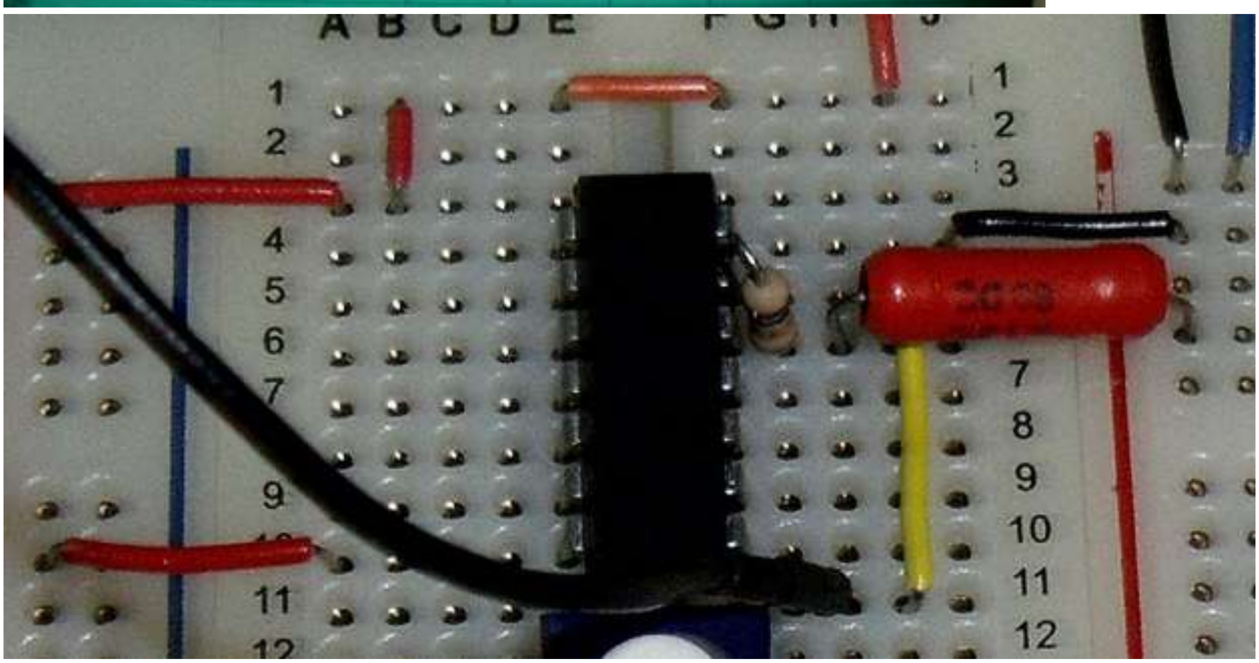
Subsystem 1 (8KHz Relaxation Oscillator):

Firstly I connected up my values, but I noticed that the frequency was not what I expected it to be.
So I changed the values around in the circuit, and managed to find a capacitor which gave me a much closer value to 8000Hz.
I then connected up the clock with the new values which while the formula did not agree with, but this brought me closest to the 8KHz mark I wanted.
We can work out the frequency of the clock by first working out the period, you can see it takes up 5 squares for two periods, so 2.5 squares at 50uS means that a period is 50*2.5 = 125uS, then you put 1/(125E-6) and you get exactly 8000Hz.
However as the oscilloscope can measure more accurately than we can see, it sees the frequency as 7.974KHz, which is very close to 8KHz.



Now if we do some calculations we can work out that the time that could be recorded on the maximum rate:
T= Max number of samples / Samples per second = 524288/7974 = 65.75 Seconds of recording and playback time.

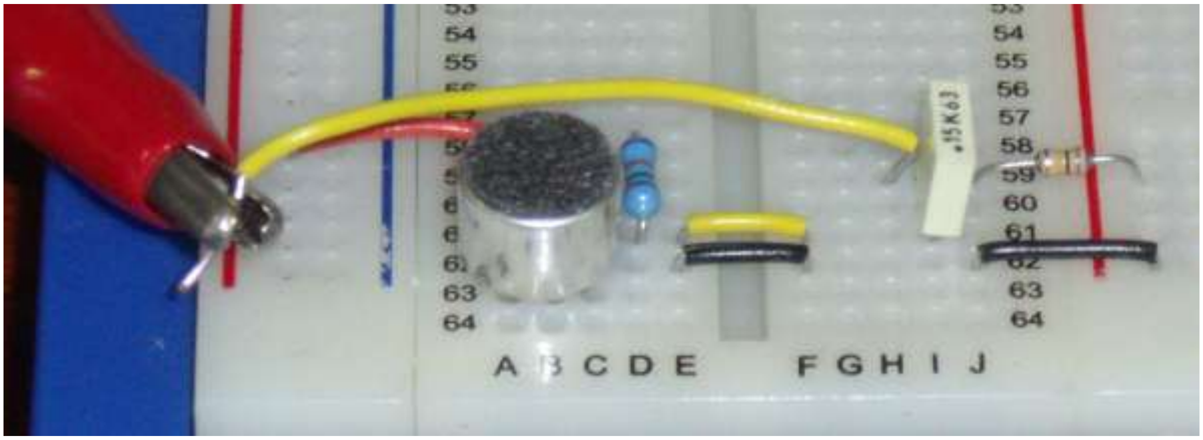

50µs
4MS
Y1 DC
21.0µs
ct-f:Y1
7.974kHz
(5*50E-6)/2 = 1/8000
period. 1/period =
frequency, so 8000Hz
Y1
Y1:2V=
rfr

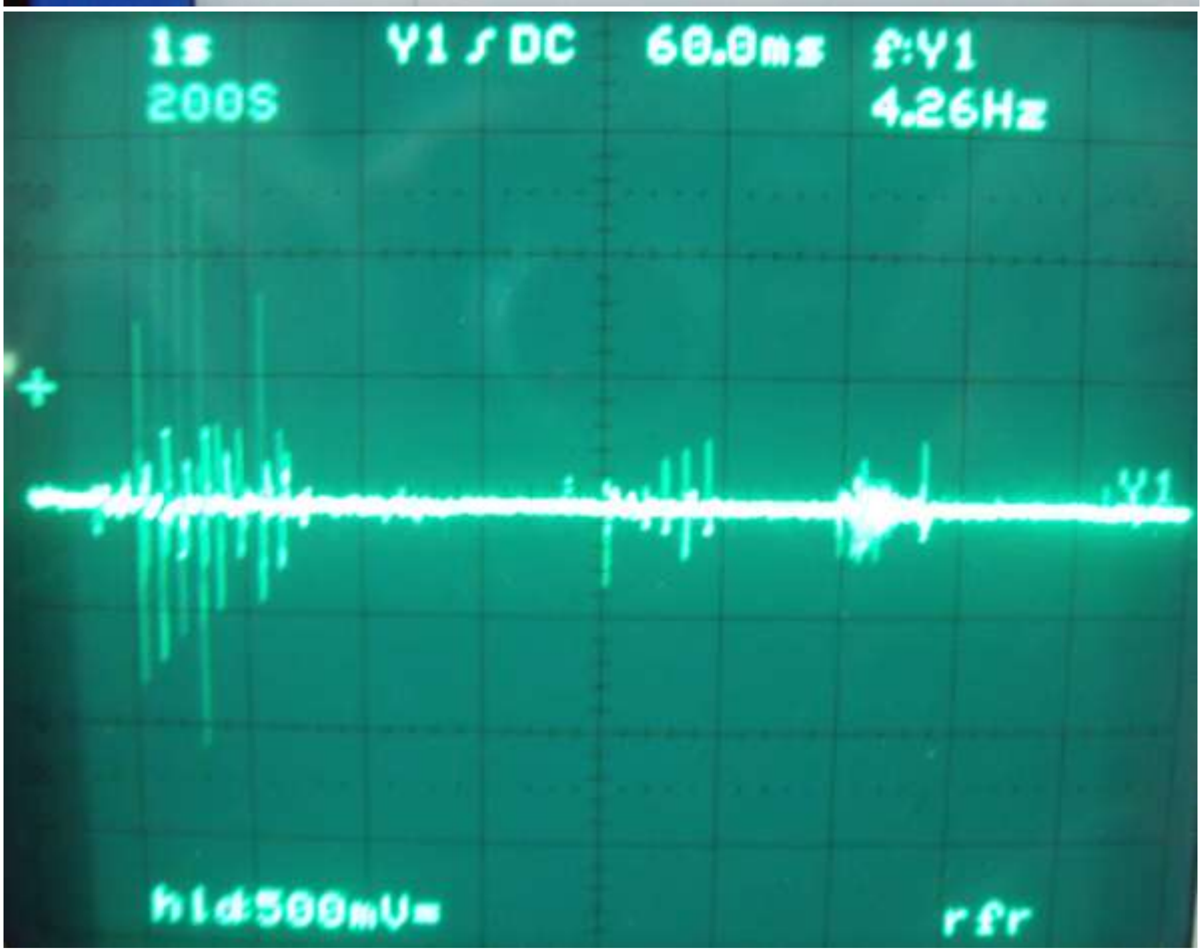


Subsystem 2 (Microphone and Preamp):

Microphone:

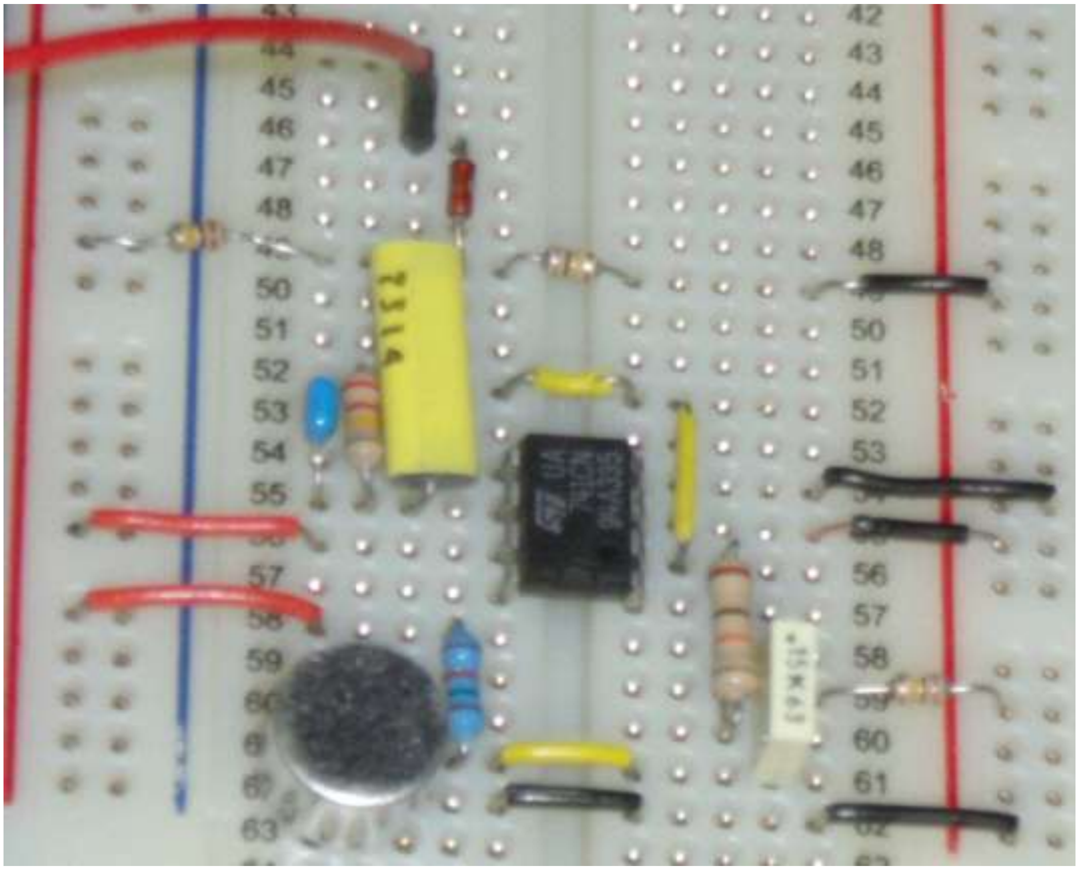
An oscilloscope connected up the microphone and got the correct output shown in the pictures when connected to oscilloscope.





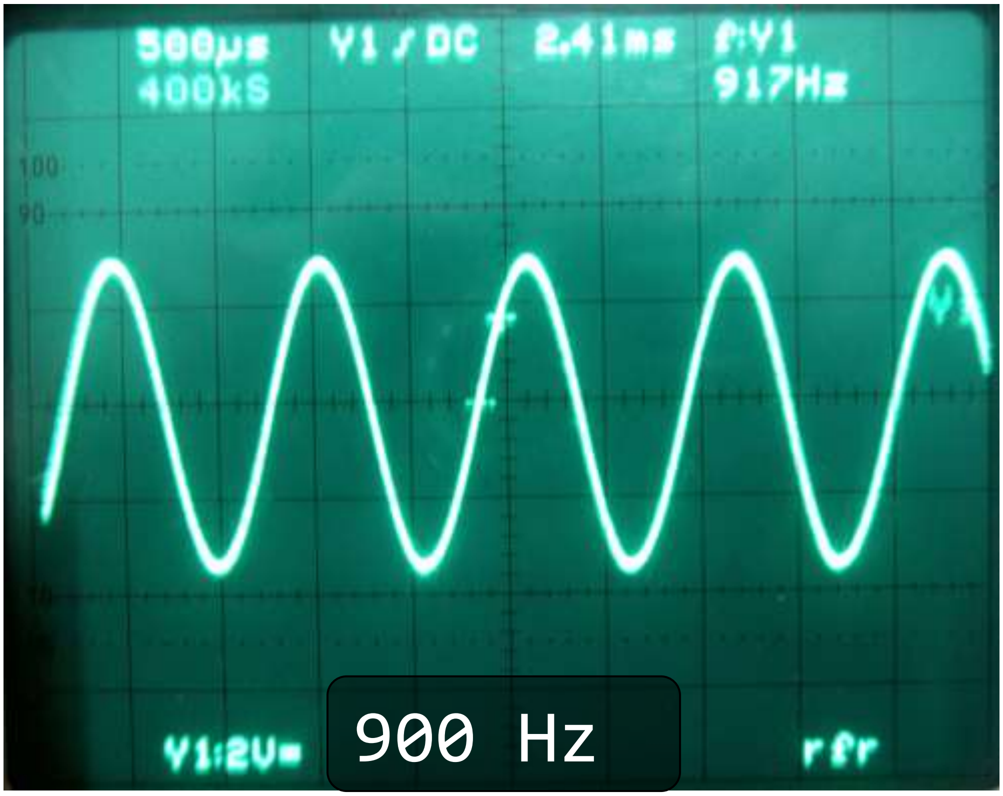
The first bumps are tapping the microphone; the two parts on the right side are speaking into the microphone. The signal works and is correct. The higher voltages are not to be worried about as they will just saturate in the preamp.

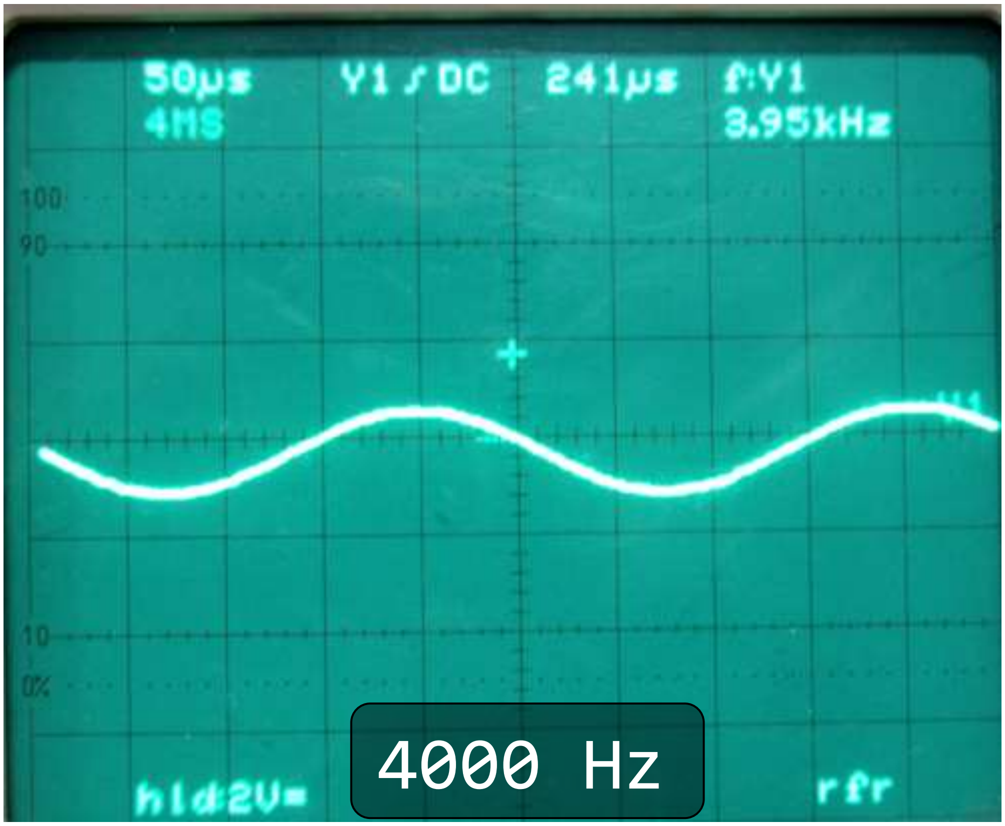
Preamp:

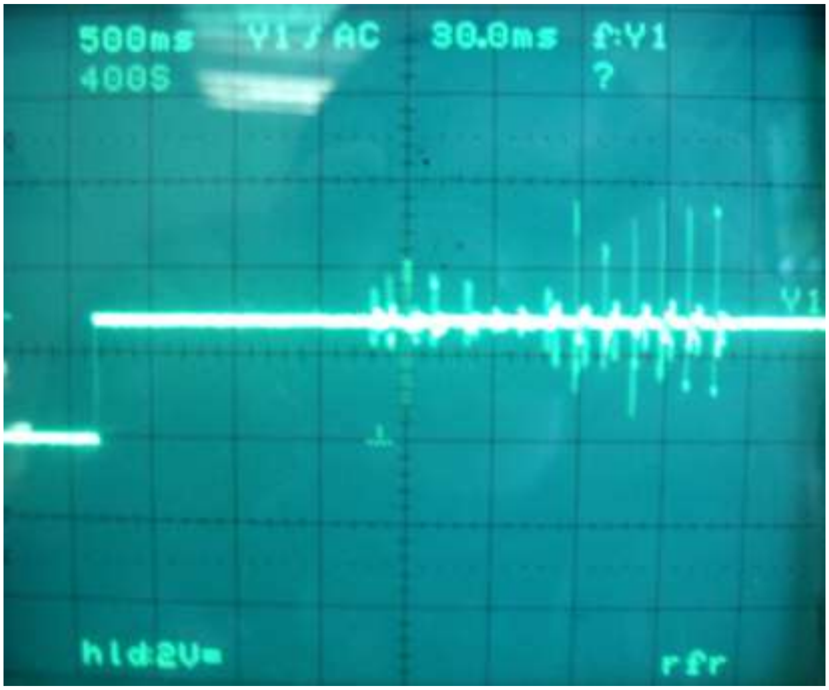


Firstly I connected the wire to before the coupling capacitor and checked the for the frequency cuts. I did this by connecting a signal generator with low amplitude (Around the same I expect from my microphone) into the preamp.

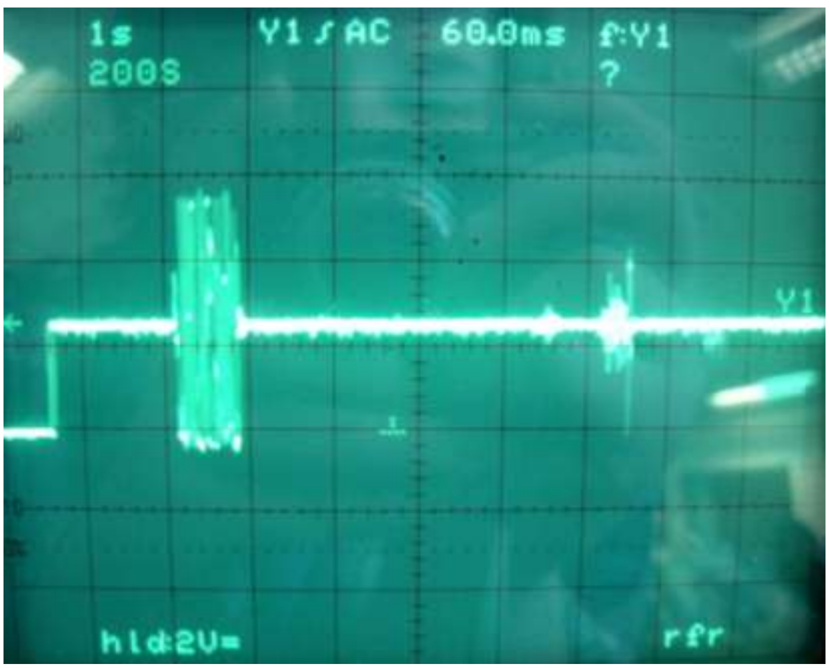
As you can see the amplitude of the 4000Hz signal has clearly been cut down a lot, this will be enough to prevent aliasing in the circuit when the signal is sampled. This also allows for all the average male human voice to be captured.
Then I connected up the coupling capacitor and the voltage offset, with a diode to prevent negative voltages.


500µs
400kS
Y1 ↗ DC
2.41ms
f:Y1
917Hz
100
90
Y1:2V=
900 Hz
rfr


50µs
4MS
Y1 ↗ DC
241µs
f:Y1
3.95kHz
100
90
10
0%
4000 Hz
rfr



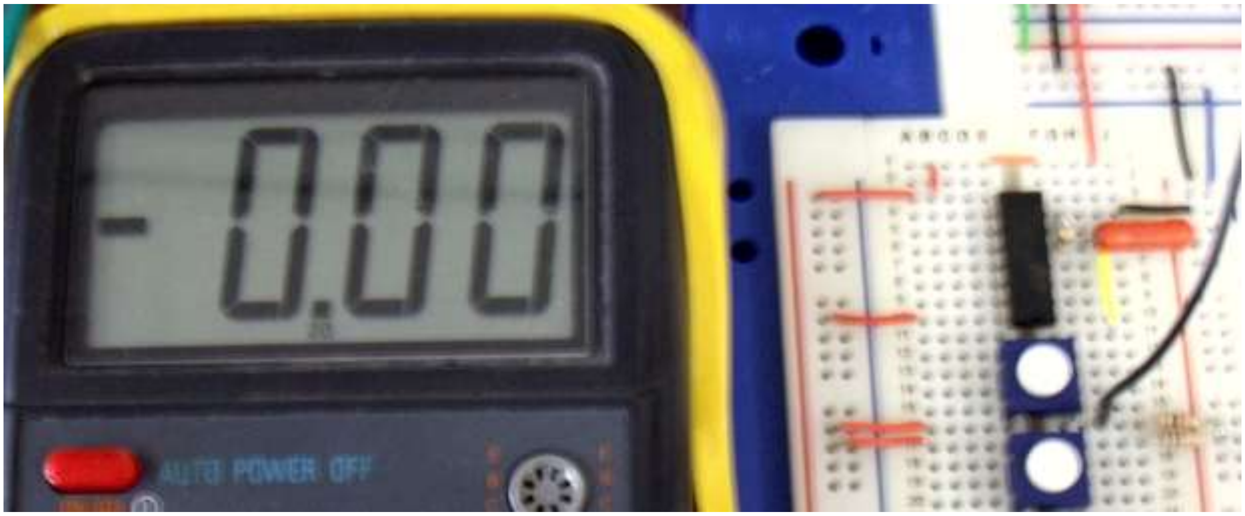
This is where I was tapping on the microphone; on the very left you can see where 0v is, before I connected the signal. The voltage offsets at around 2.4v which is about right. The slightly lower is probably due to the diode taking more than 0.7v

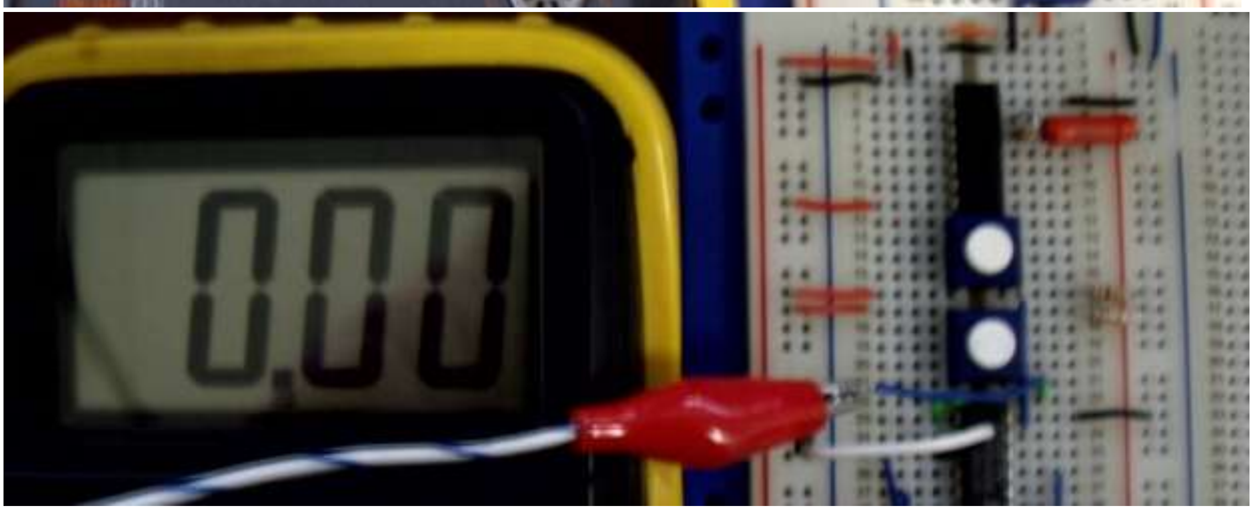


This is where I made loud noises to test the limit. As you see the chip will not go much below -0.4V, so the chip cannot be damaged, as well as this it will not go much above 5 volts so the chip is safe.

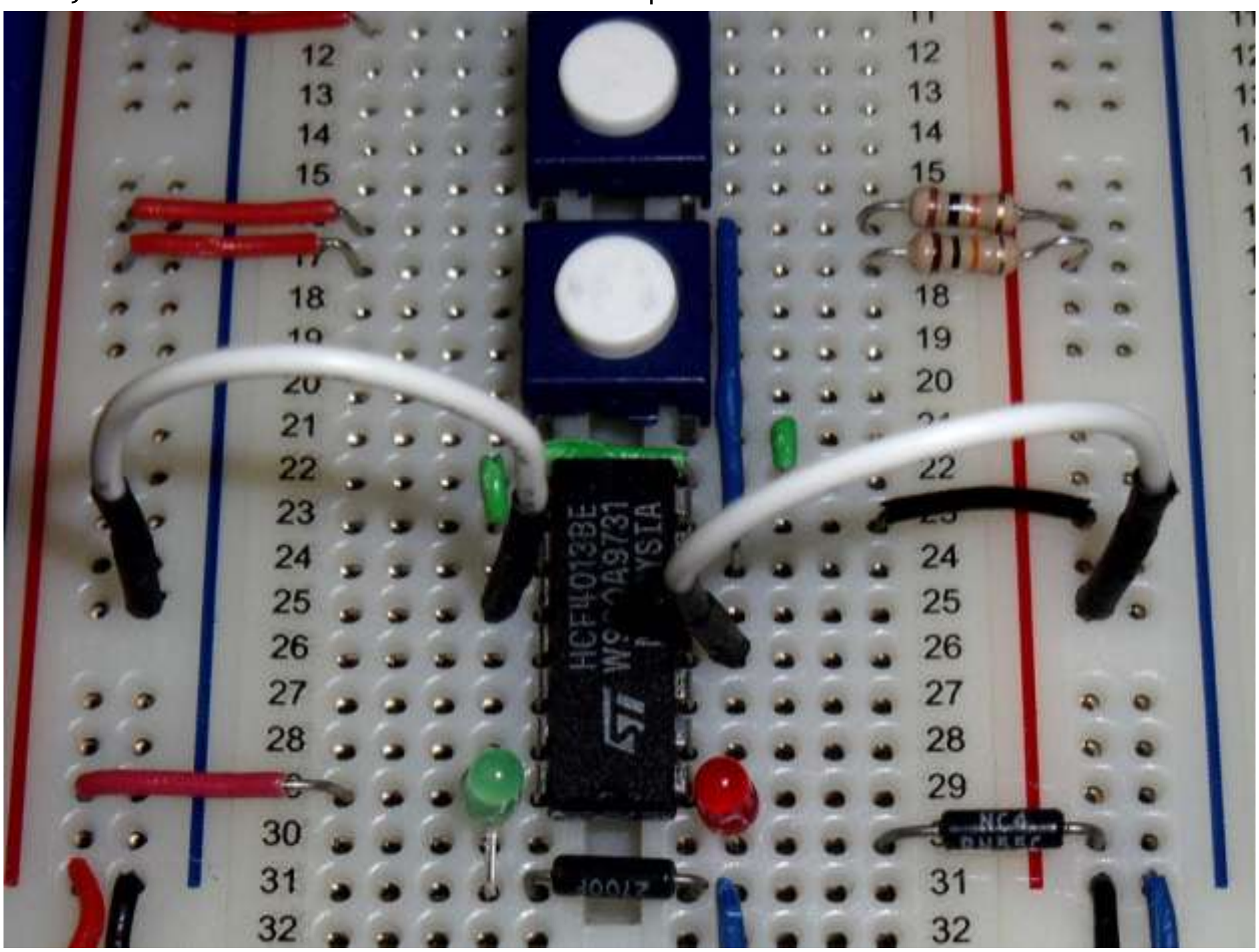
Subsystem 3 (Control Lines):

First I connected up the hold down; push to make switches which when pressed should give a high signal and when not give a low signal.

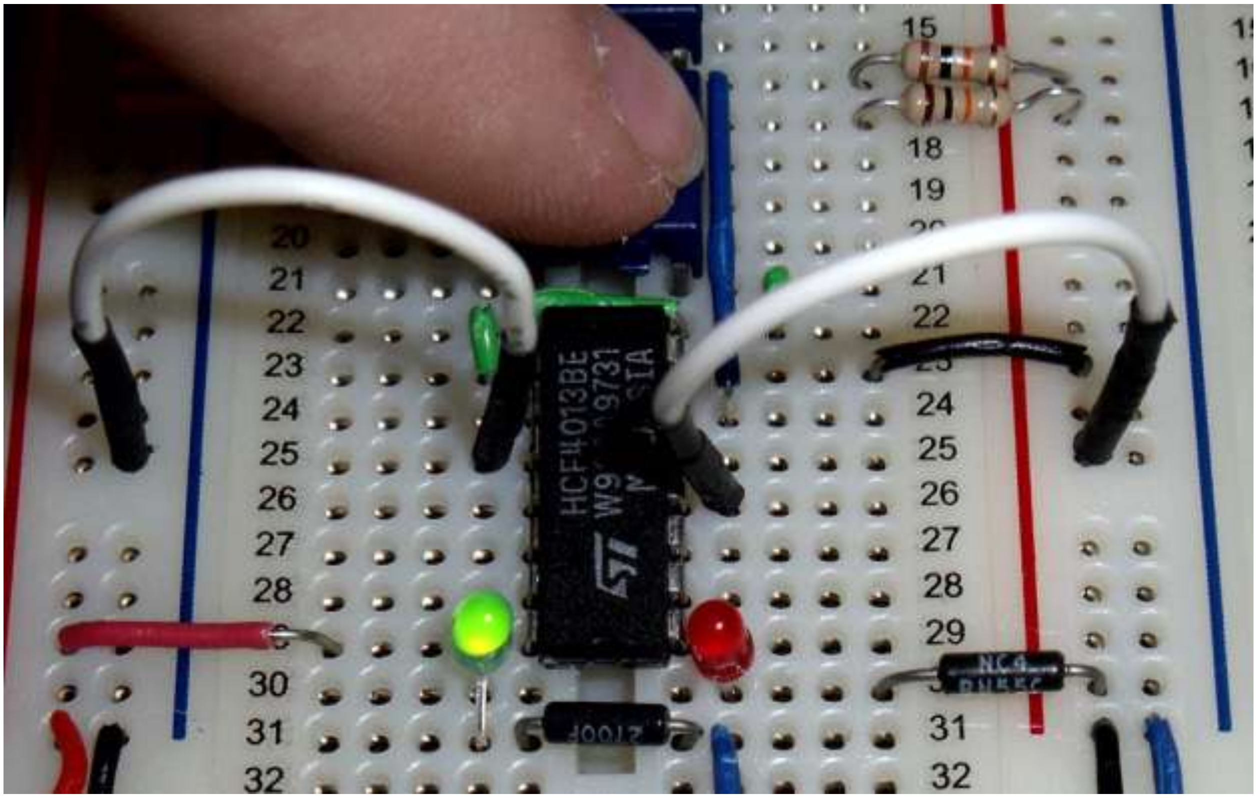




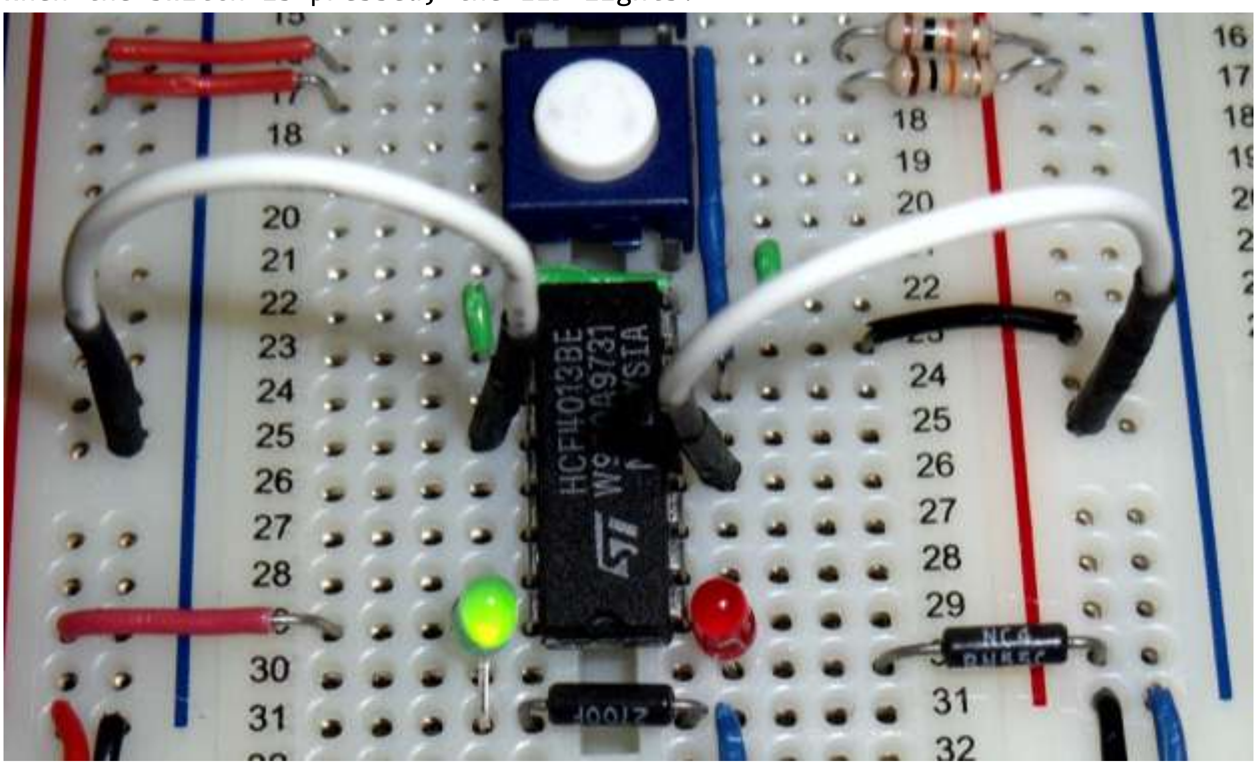
This clearly shows the switches working, and then I connected up the latches. With LEDs on Q and the resets connected via jump lead to 0v and 5v for when used, these will be later connected to specified line on the counter:



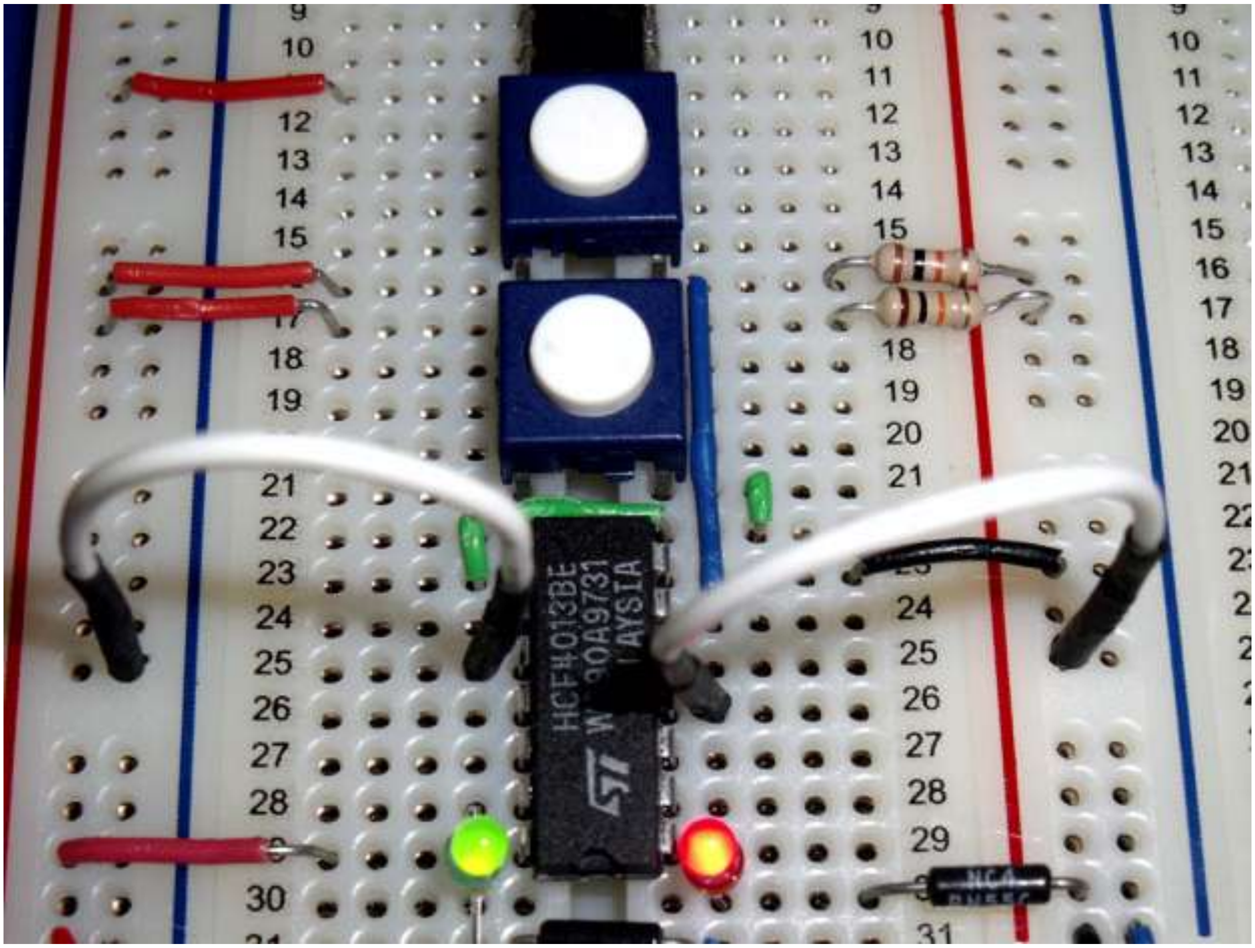
When no switch has been pressed, Q remains low.



When the switch is pressed, the LED lights.



And then the LED remains on until the reset line is put high.



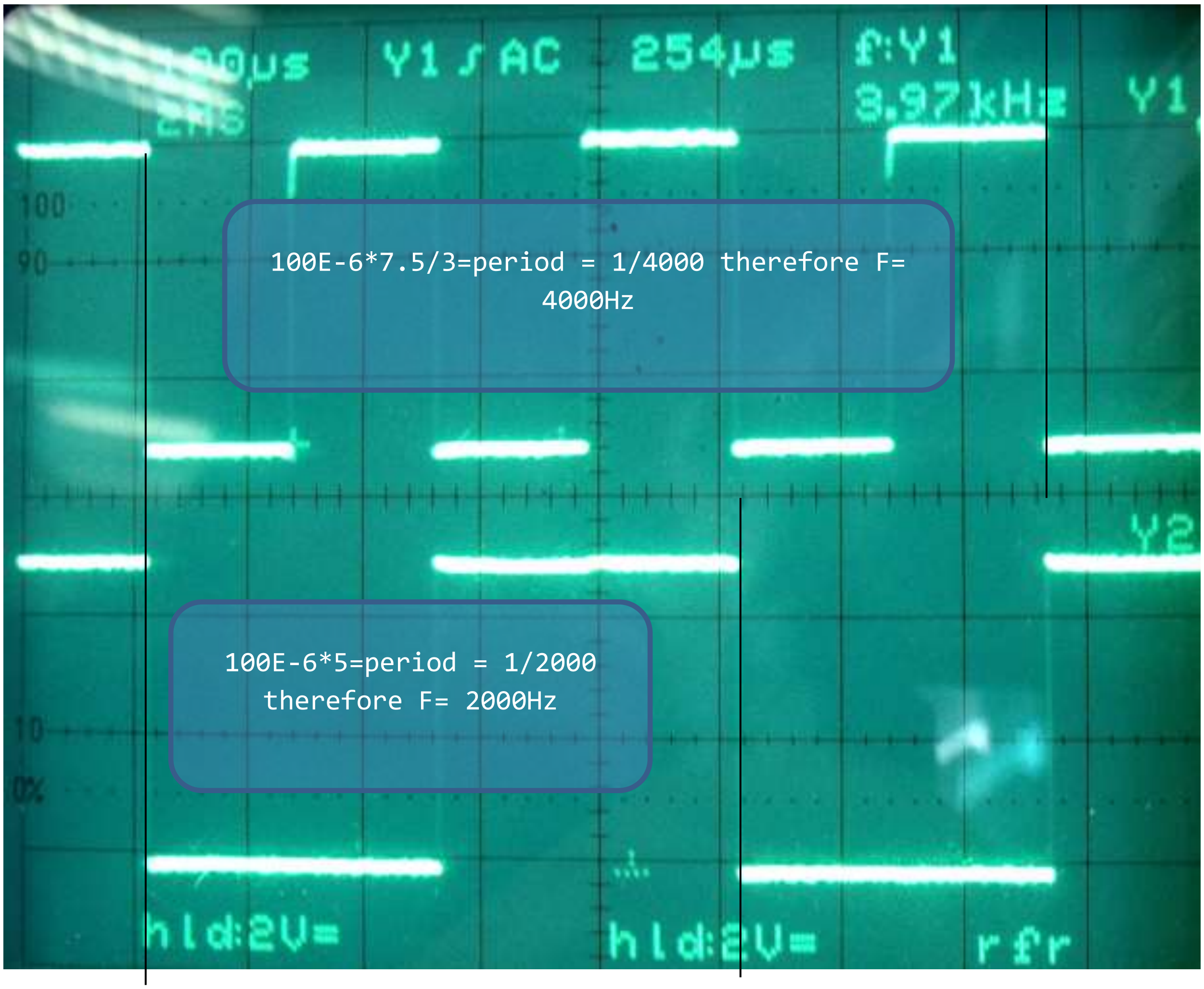
The same works for the second LED. Then I connect each of the jump leads to 5v and back and they both return off.

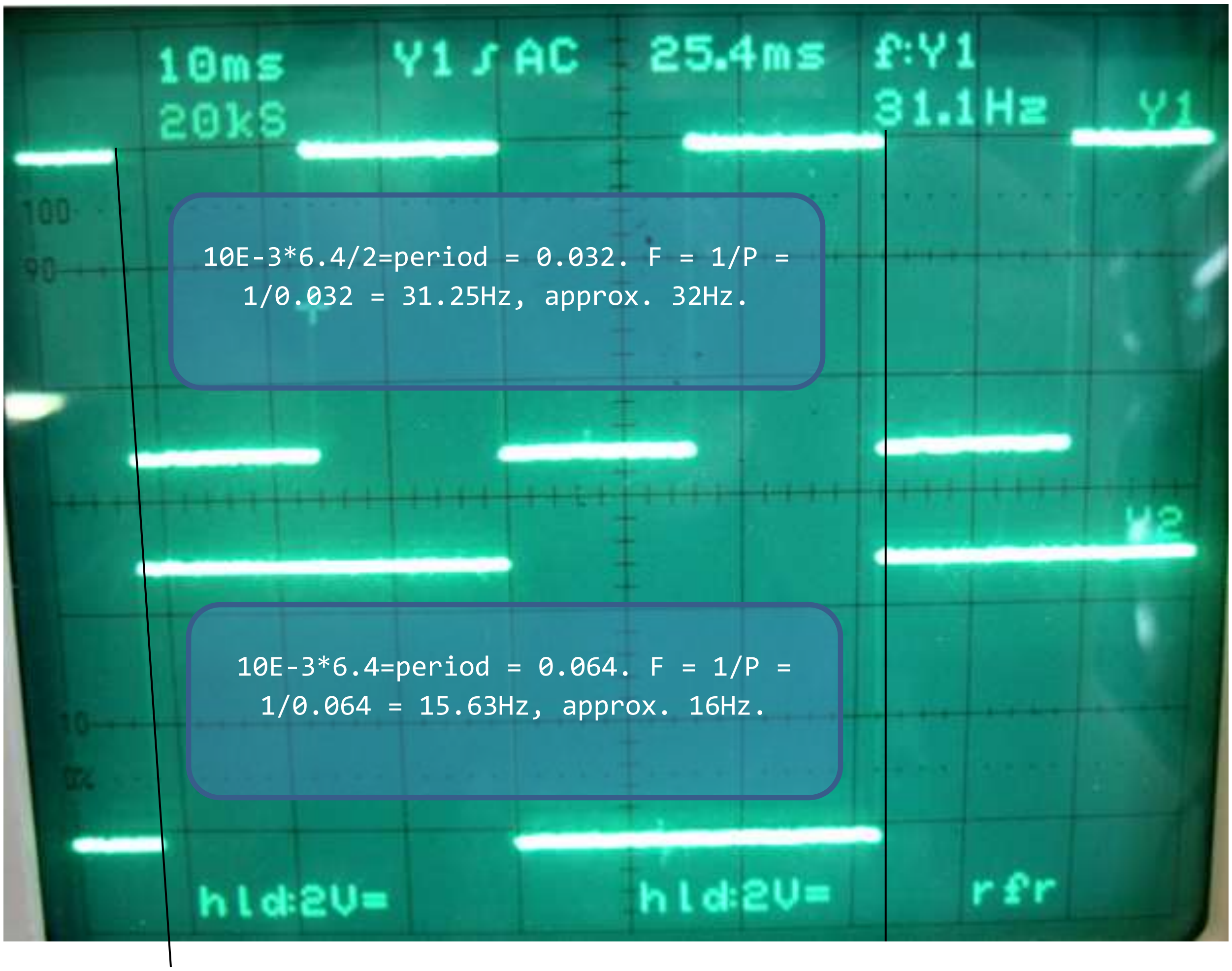
Then I connected up the three NAND gates to the output of this, as well as connecting the resets together and placing it so it is easy to receive signal from the next subsystems. I test it so that when no latches are enabled, OE# and WE# are high so that the chip is not outputting or recording. But when the record button is pressed the WE# line goes low, and the AS line goes high, and then reset to high. And when the Listen button is pressed the OE# line goes high but the WE# stays high and the AS remains low. If both buttons are pressed then WE# and OE# lines remain high and AS is low.

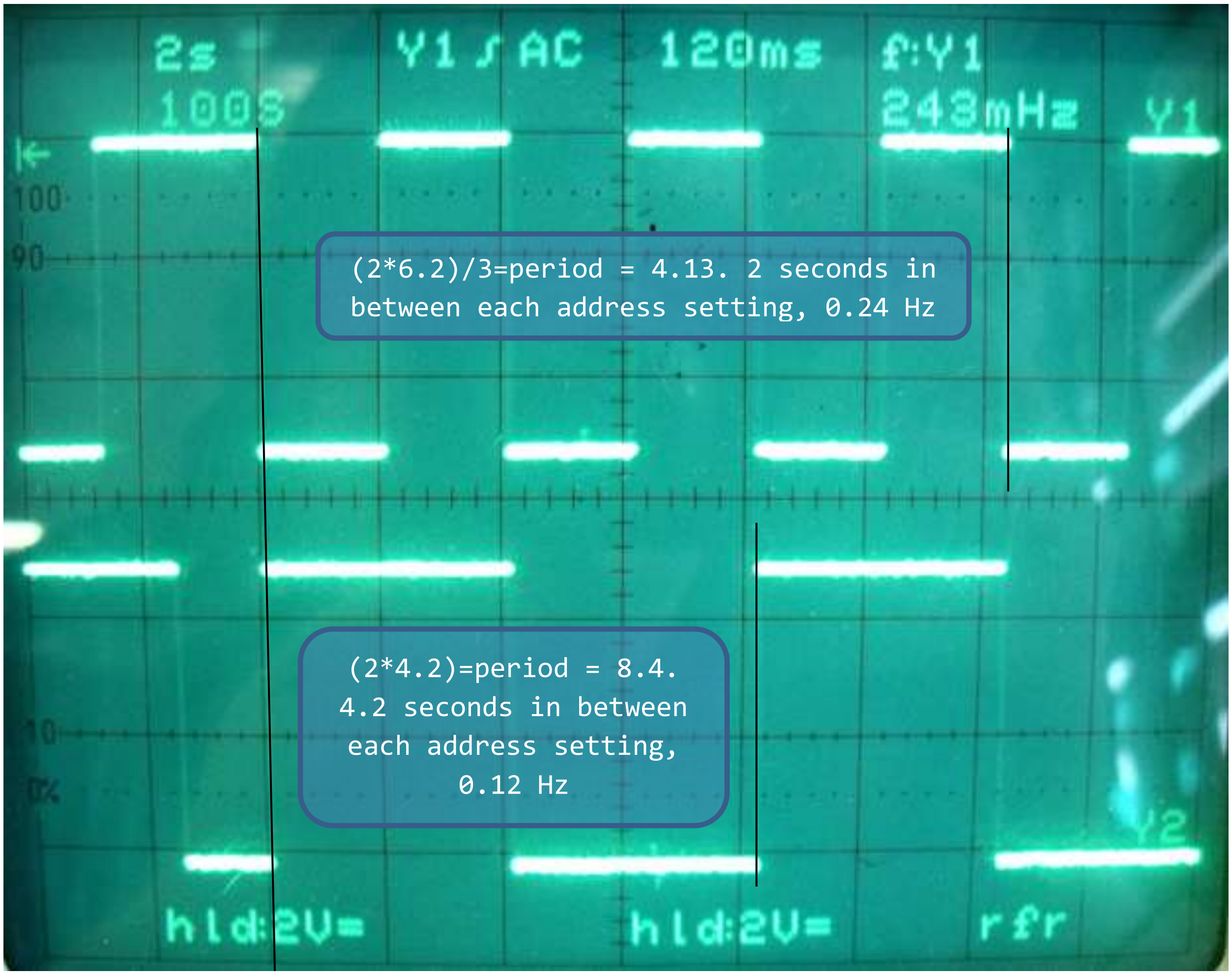
When testing this I realised that I had got the LEDs the wrong way around, so I swapped their positions and then it indicated the right channel.

Subsystem 4 (21 Bit Binary Counter):

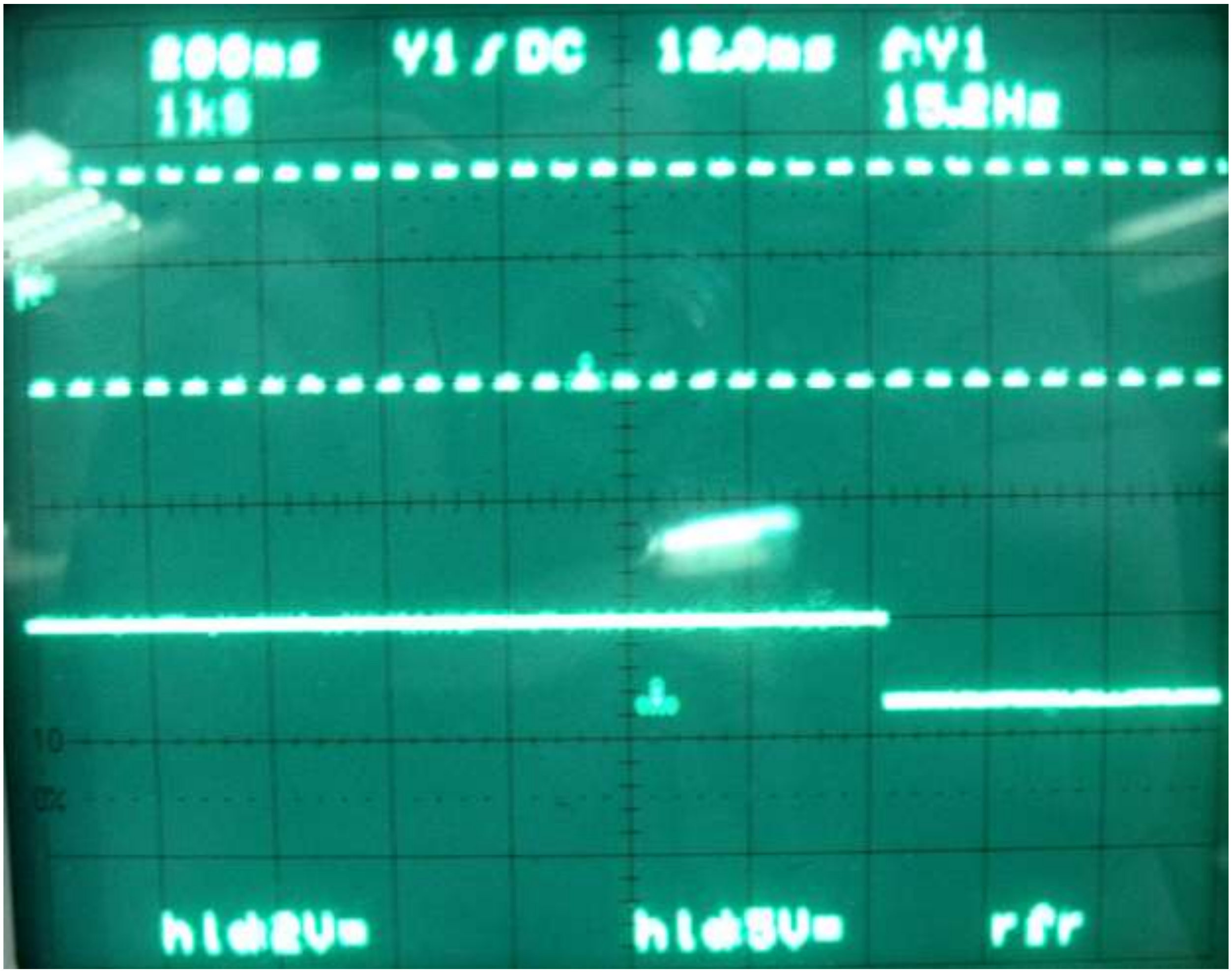
I connected the first and second data line of each chip to a bar graph display and also to the oscilloscope. I can see the first two of the first chip being too fast to see switch on and off, so I connected them to an oscilloscope to check that they were. The second two were flickering very fast and the last two were slowly changing between them as if it was counting up. In each Period there are two addresses, one where the bit is high, one where low. So each frequency should be doubled for the number it counts as. So the first lines frequency should be 4000Hz, but really it is 8000 changes a second.



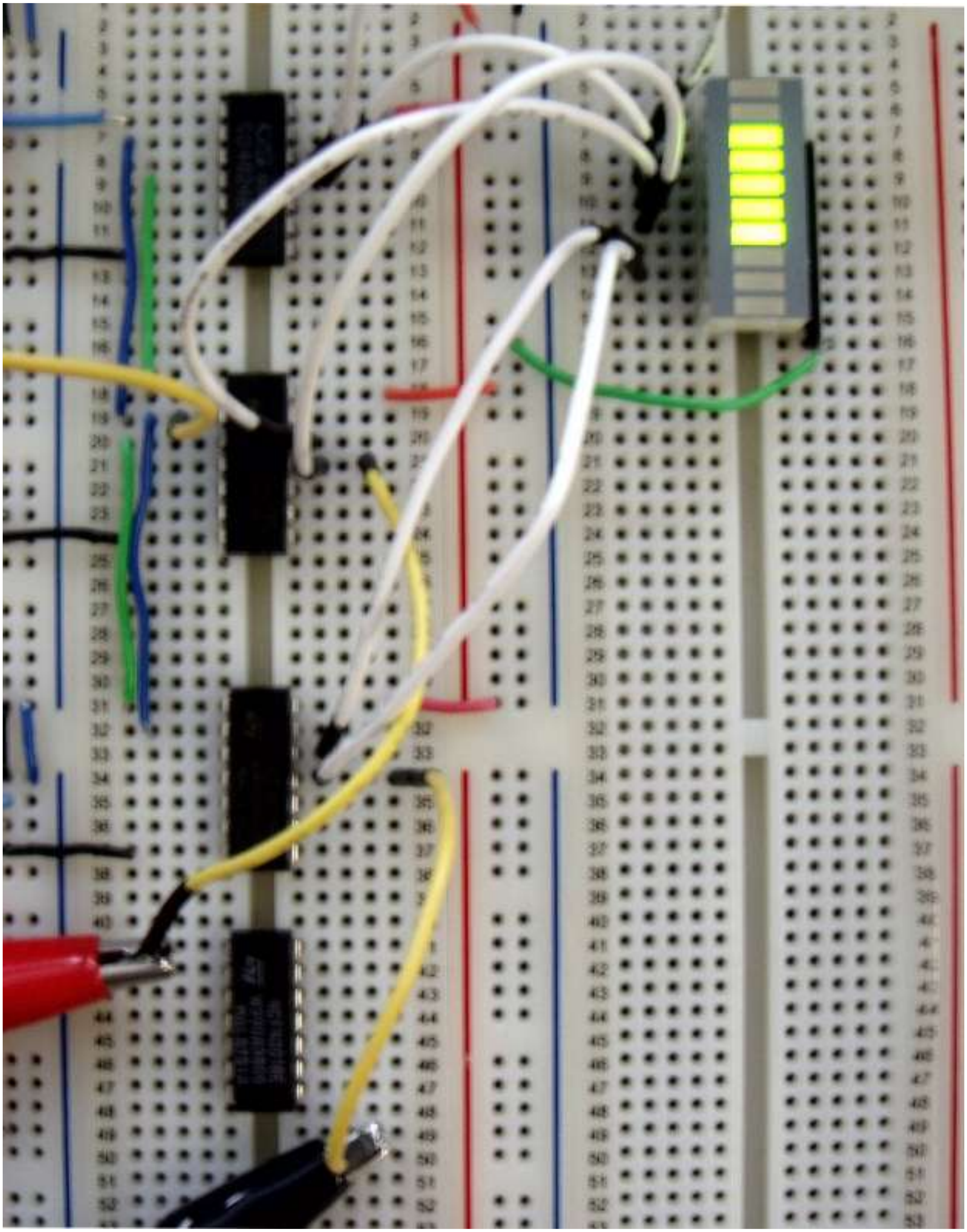

10ms
20kS
Y1 AC
25.4ms
f:Y1
31.1Hz
Y1
Y2
10E-3*6.4/2=period = 0.032. F = 1/P = 1/0.032 = 31.25Hz, approx. 32Hz.
10E-3*6.4=period = 0.064. F = 1/P = 1/0.064 = 15.63Hz, approx. 16Hz.
hld:2V=
hld:2V=
rfr



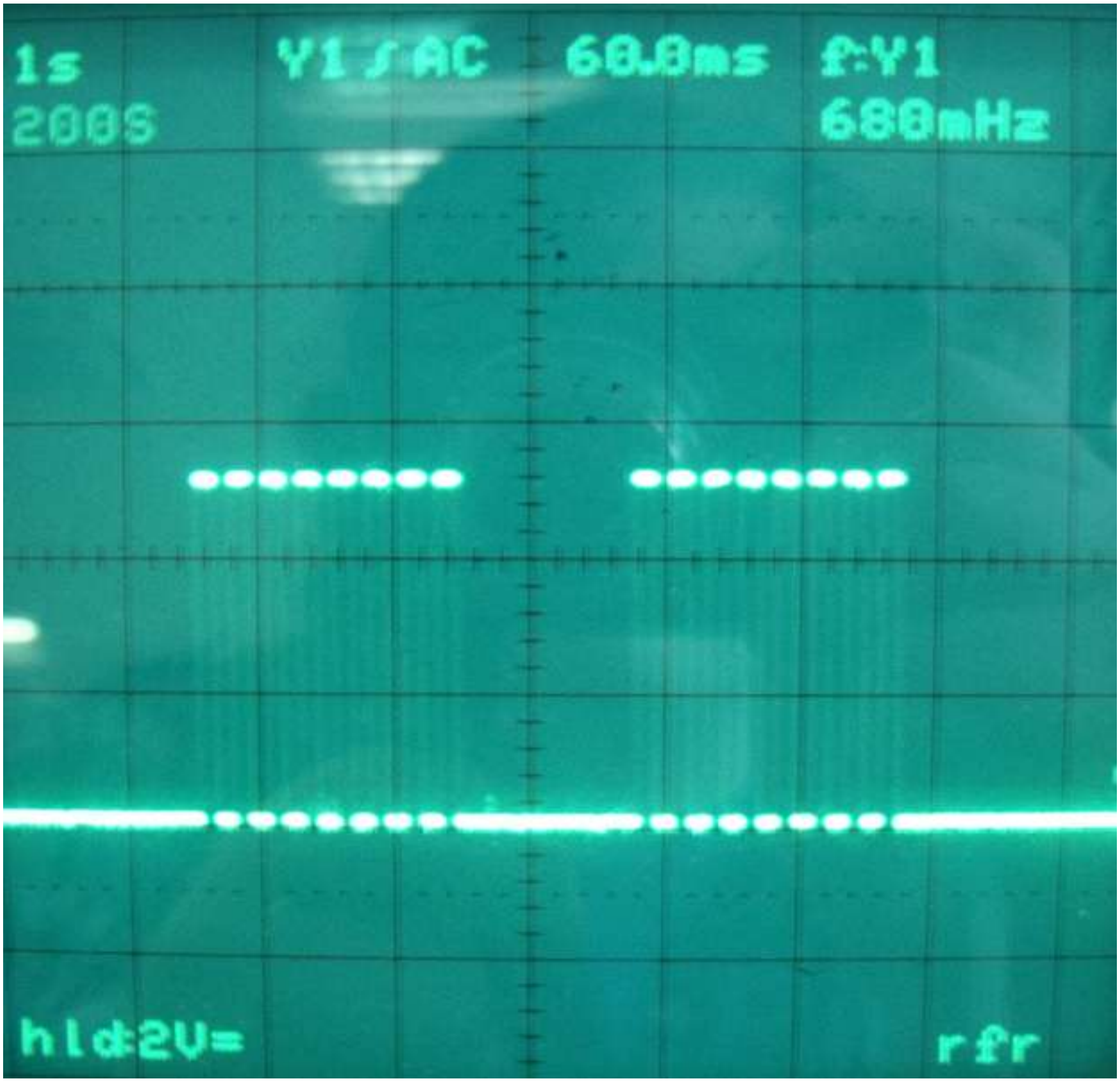
I also measured the second line of the final chip to be approximately two seconds.



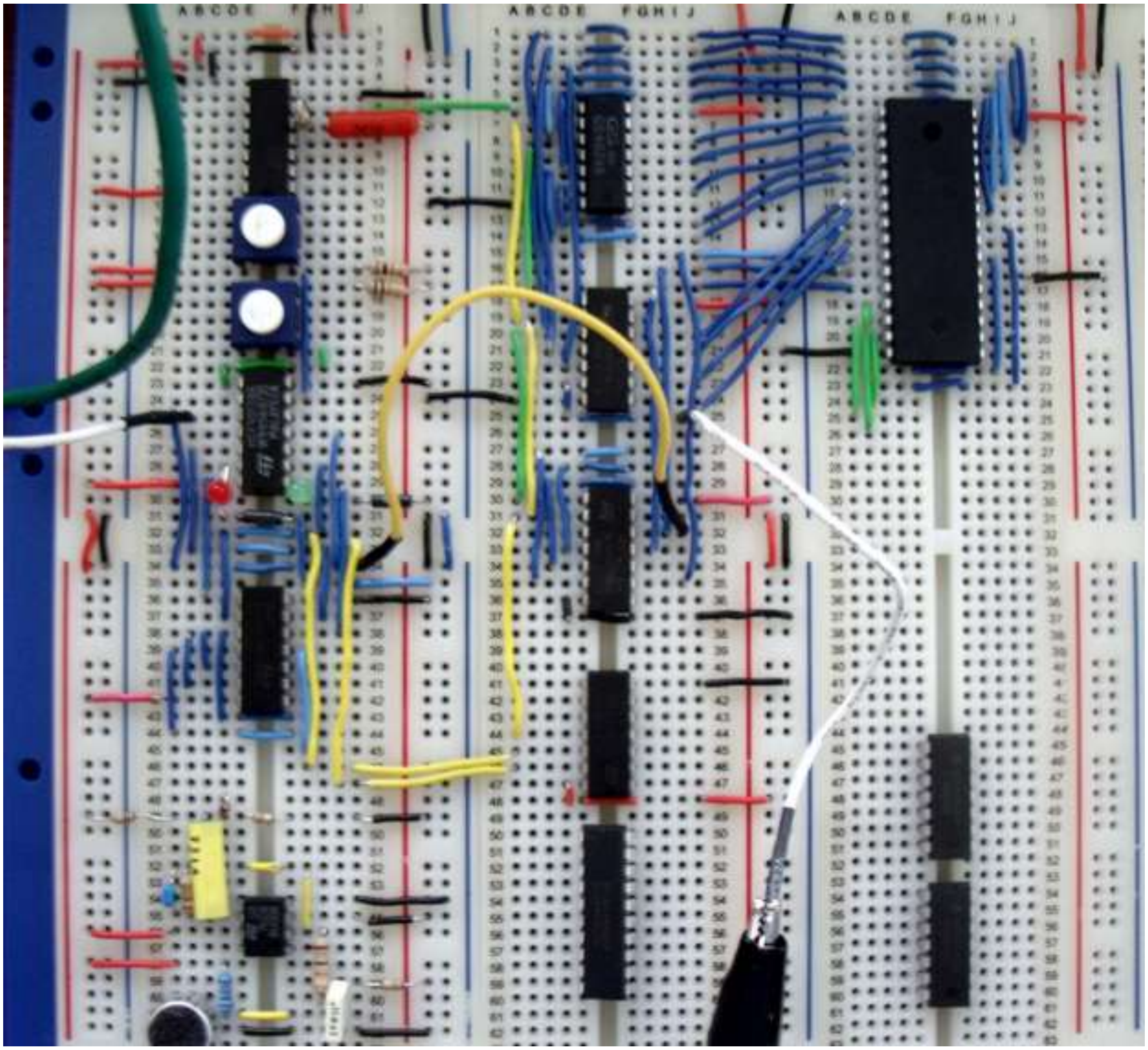
The top line is the second line of the second chip, and the bottom line is the second line of the third chip. The top line is going at approximately and the bottom line is going at approx. 2 seconds. All these numbers are exactly as planned and hoped for.



Then I connected the reset line up to specific points on the counter and checked. I also connected up the AND gate on the input to the reset on the counter. I then tested pressing each button and seeing the signal count up for the time until reaching the point, and then stopping until the other or next button is pressed. I set the timer for the first line of the 3rd counter. This lasts for approx. 2 seconds.

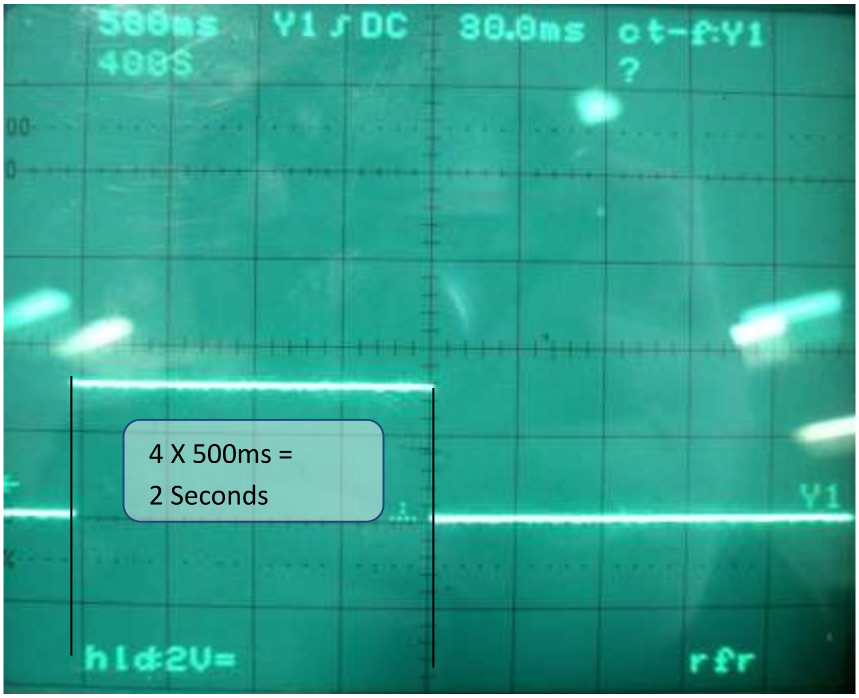


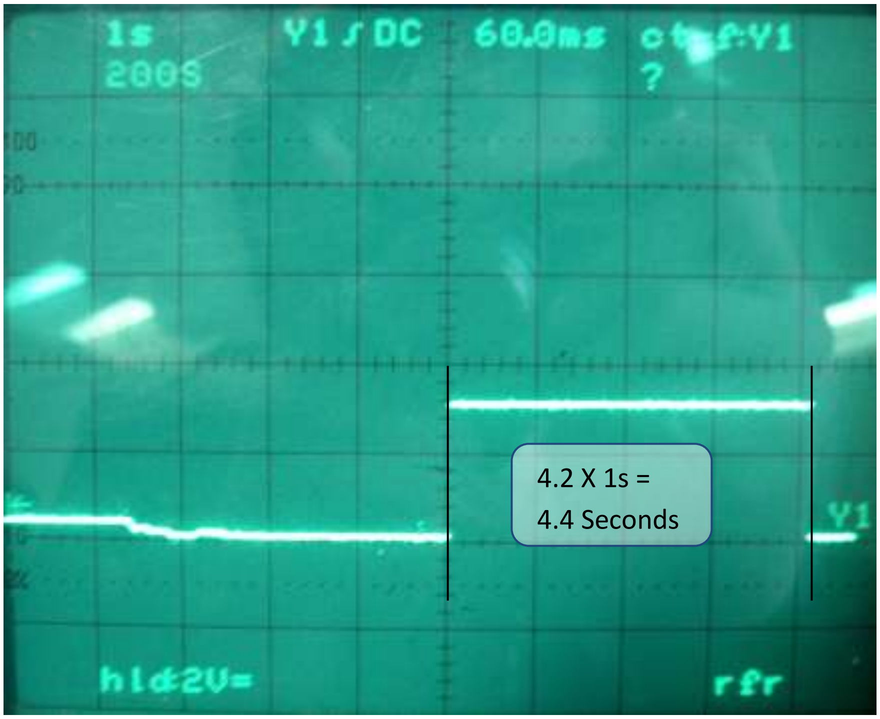
The counting wire is set to the 4th line from the second counter.
Seeing this successfully work I now know that my control lines and counter work perfectly with the record and playback time set-able by moving a jump wire.

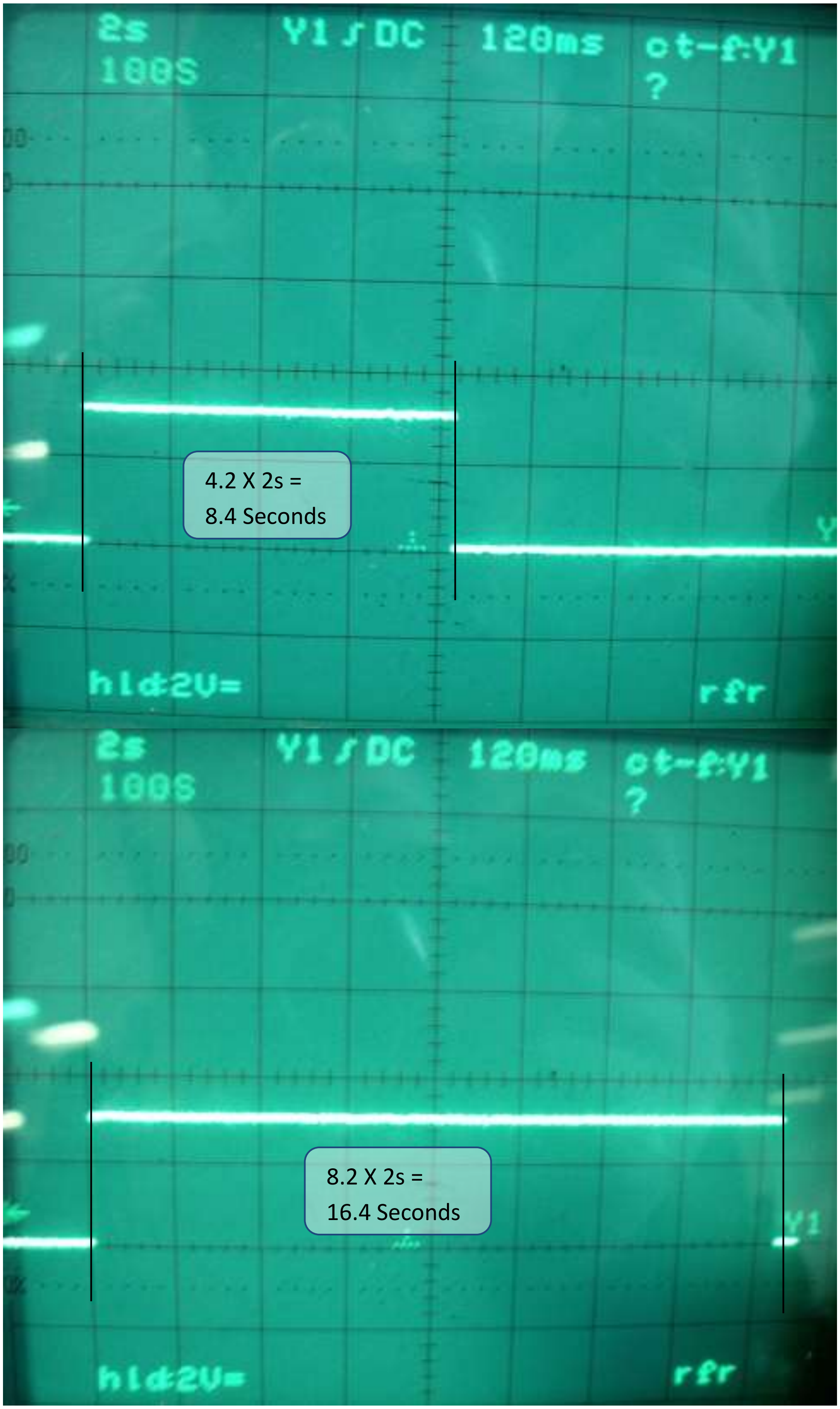


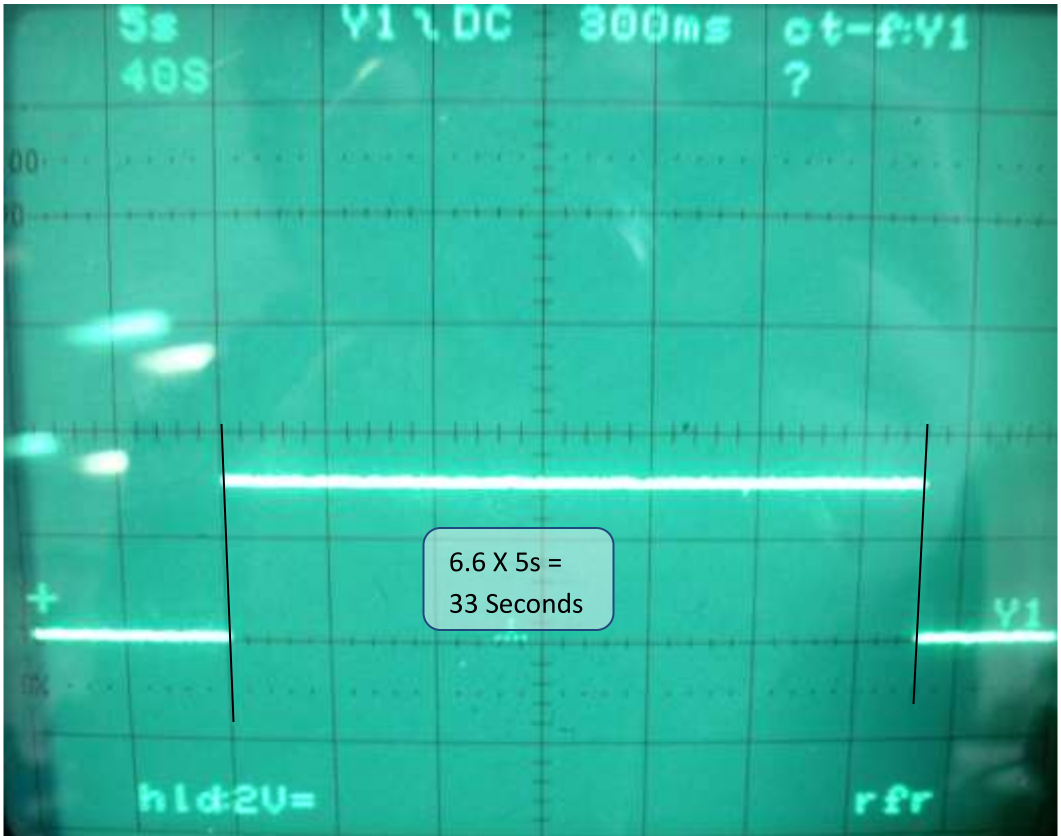
The next group of photos all show the pin going up in significance from the final counter. Starting at the First bit on the chip, going up to the last one that is usable for my circuit.

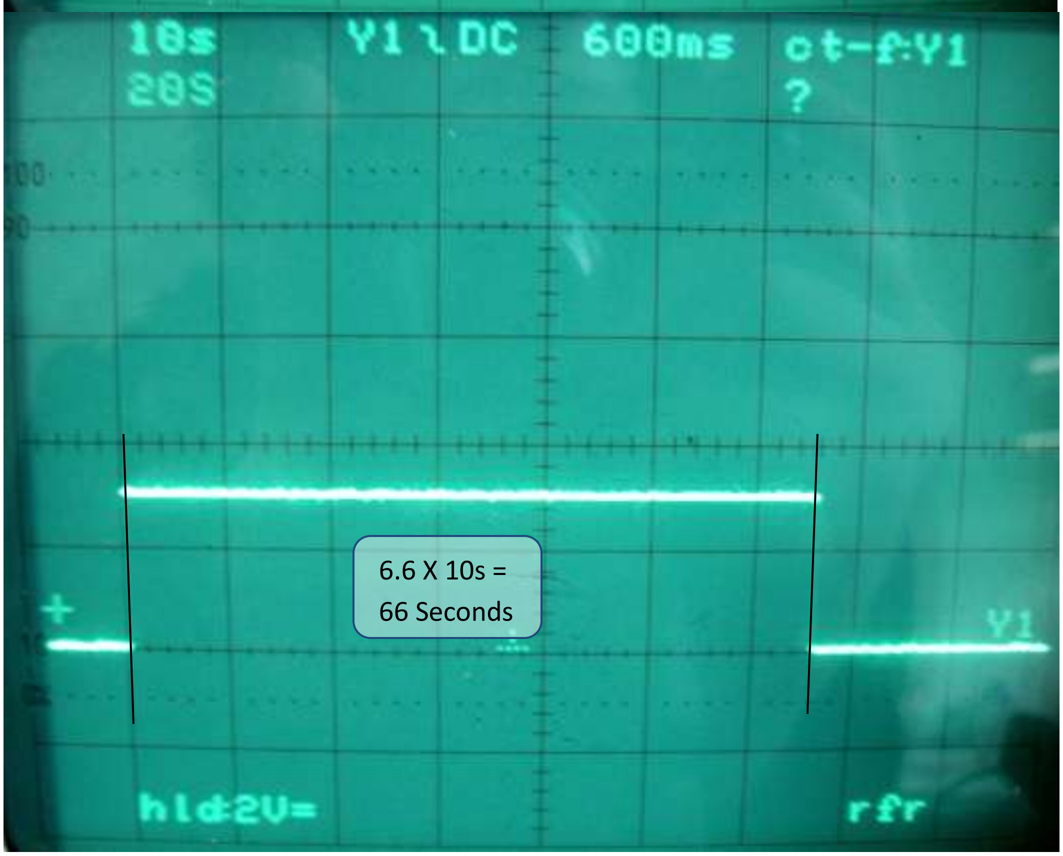
As you can see, they all fall around the estimates that I predicted with some of Them going over is due to the frequencies not being exactly the same as the time. While you half 8000Hz so many times, you get 125Hz, but then if you continue till the 4 seconds, it is 0.244Hz which is 4.096 seconds, so it's not meant to be exactly 4 seconds. As well as this the frequency is not exactly 8000Hz.





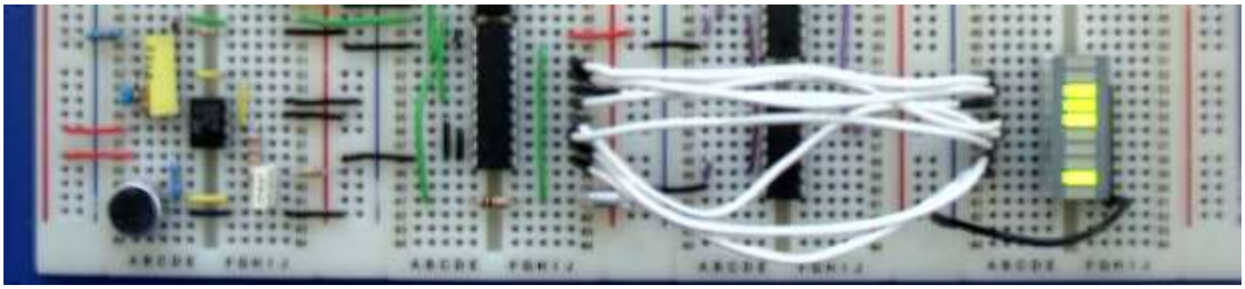

2s
120ms
4.2 X 2s =
8.4 Seconds
2s
120ms
8.2 X 2s =
16.4 Seconds


5s
40S
Y1 DC
300ms
ct-f:Y1
?
6.6 X 5s =
33 Seconds
Y1
hld:2V=
rfr

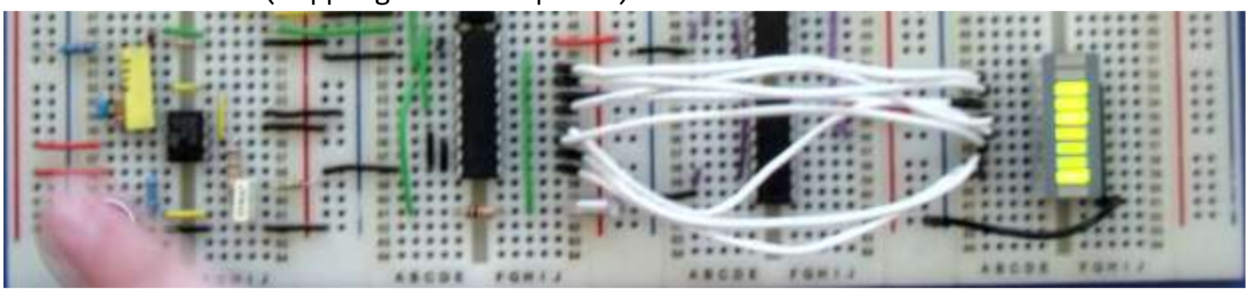

10s
20S
Y1 DC
600ms
ct-f:Y1
?
6.6 X 10s =
66 Seconds
Y1
hld:2V=
rfr

Subsystem 5 (8 Bit Analogue --> Digital converter):

Then I connected the ADC up:
This is when the sound levels were quiet; it shows that while quiet that the signal is a bit over the middle of the ADC. In reality you can see all the lights slightly flickering as they go up and down. There are still lights on when no noise which is what we want because it goes up and down and the middle voltage should be 256/2 = 128. In the picture you can see it is 128+8+4+2=142, this is because it was not perfectly quiet in the room as well as general noise off the microphone and the diode may be taking slightly less than 0.7Vs off the voltage offset.



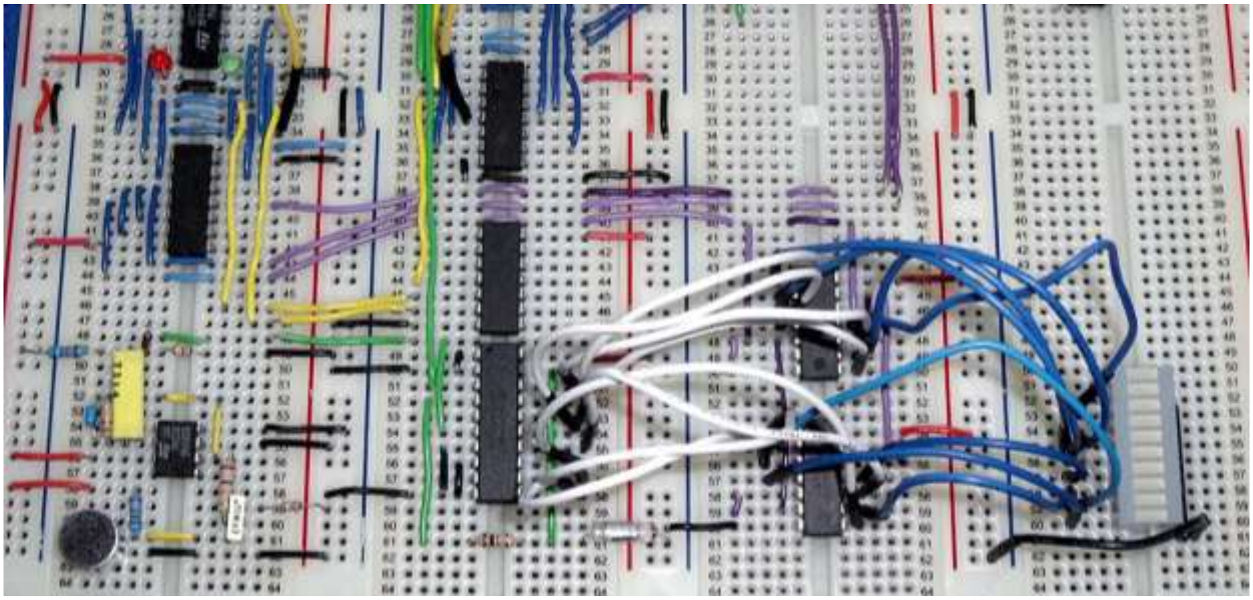
Here is it loud (Tapping on microphone)

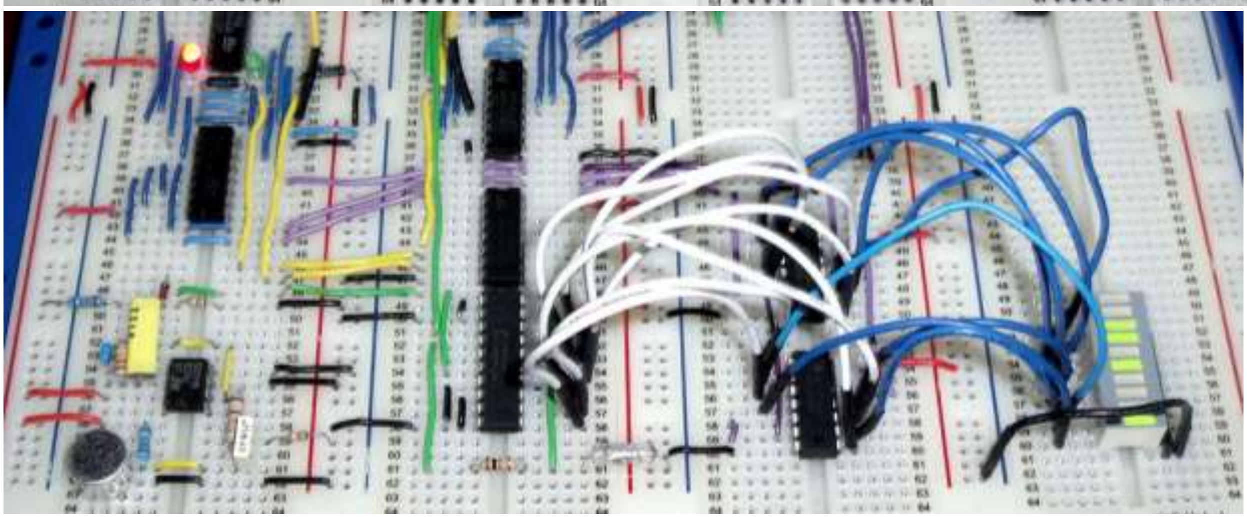


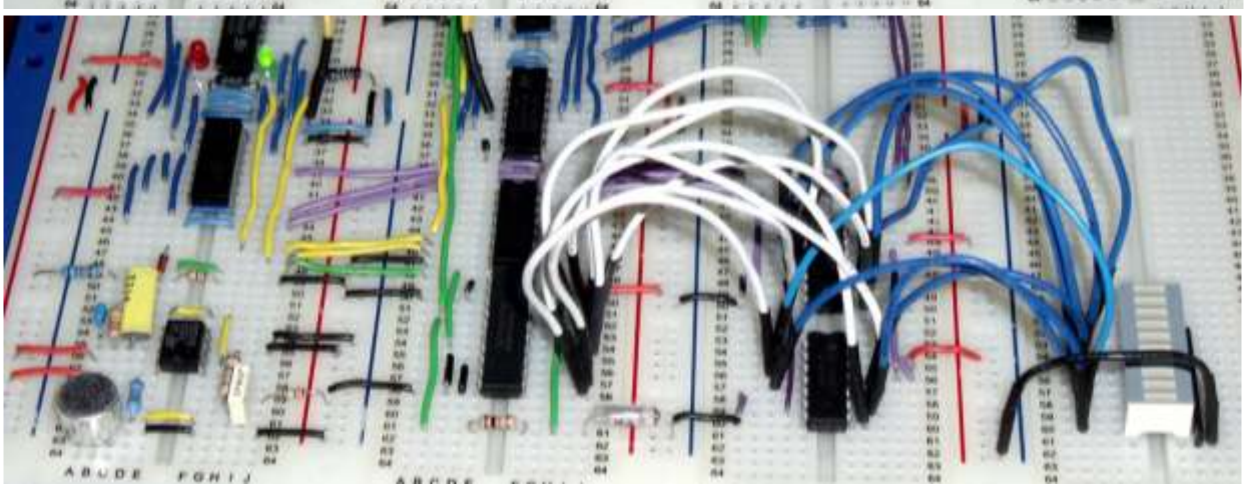
At this point there would be a high distortion on the sound but it is only tapping the microphone not actual voice that it does not matter that it is at the very peak.

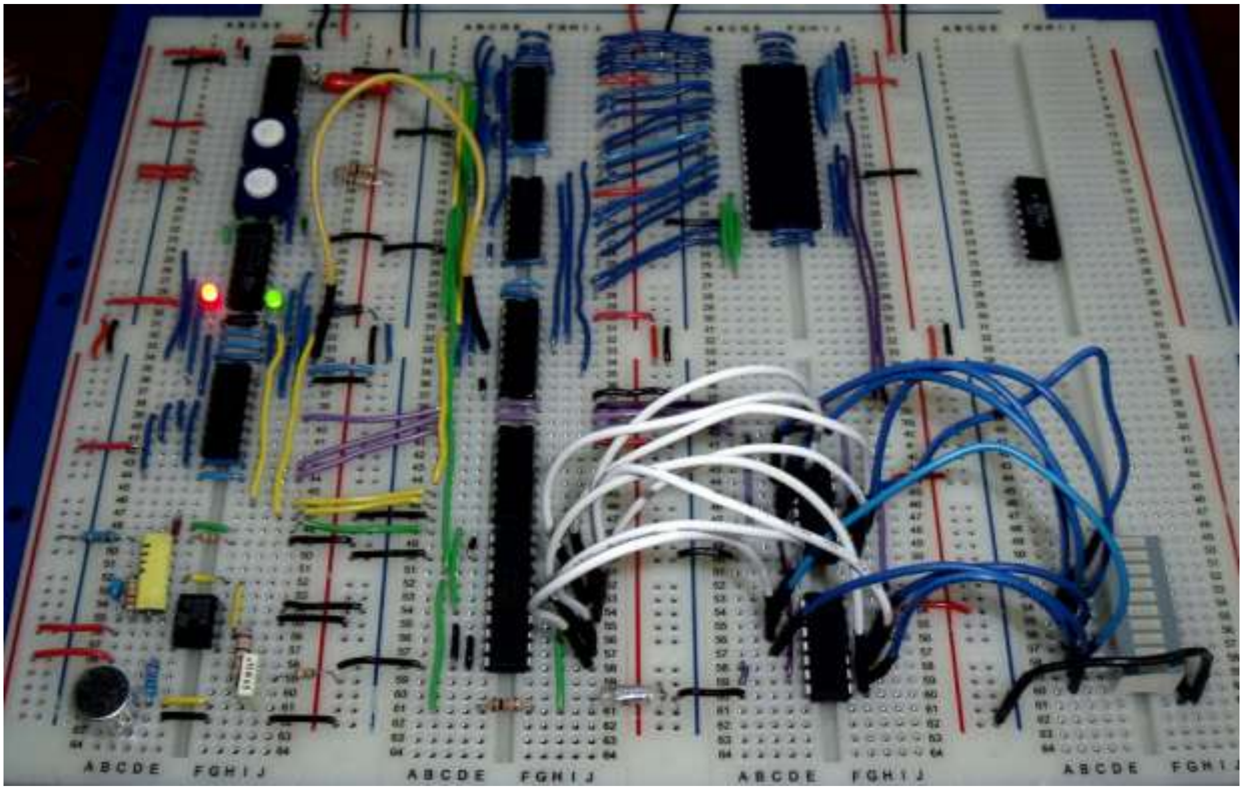
Analogue switches:

Next I connected these lines through the Analogue switches which are connected to the lines, to make sure that this works effectively.

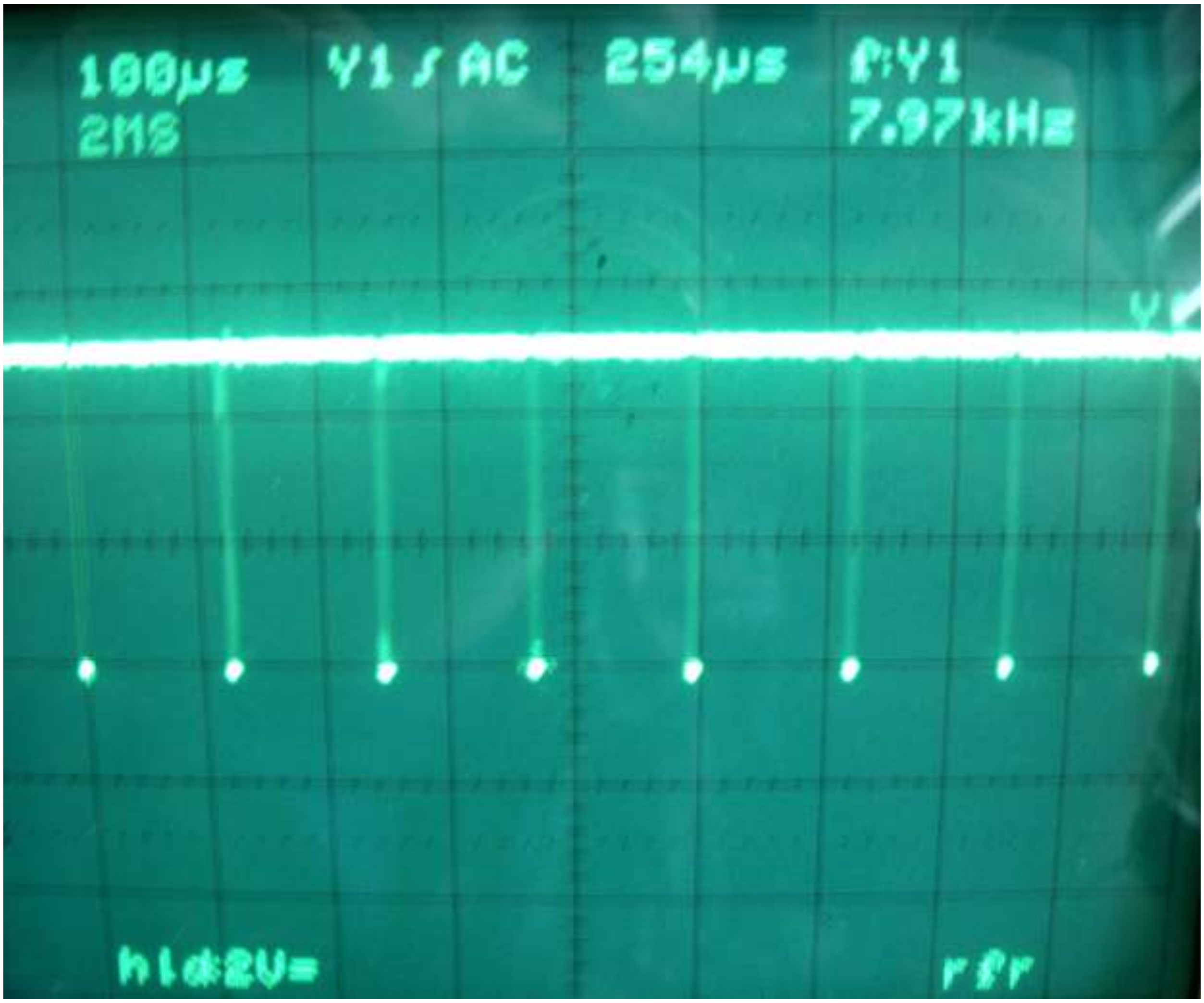








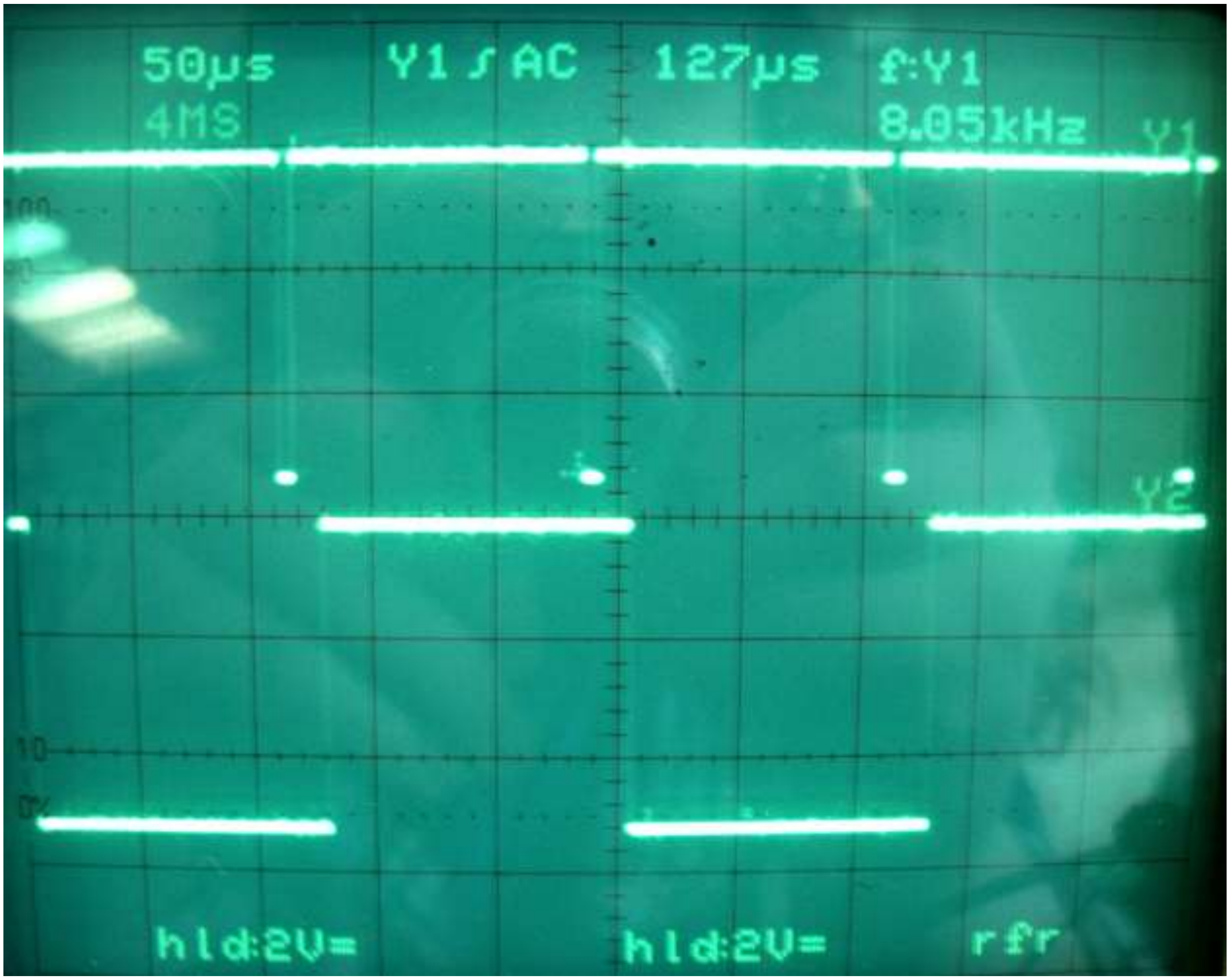
As you can see the ADC only connects to the output line when it is set to Record mode. Any other setting and no data is carried out of the analogue switches and they are left floating. This means that while playback/idle there will not be a signal on the IO bus, which could cause issues if there is both voltages being inputted and outputted at the same time.
Next I connected up the WE# line with the logic gates from the INTR# Line.



As you can see the WE# line is high almost always, apart from the pulse in between the conversion being ready and the next one starting. This is when the sample is taken from the ADC and stored into the memory. It is also very close to 8KHz which is what I wanted.

When any combination other than the only switch being latched is the record, any other and it is held high.

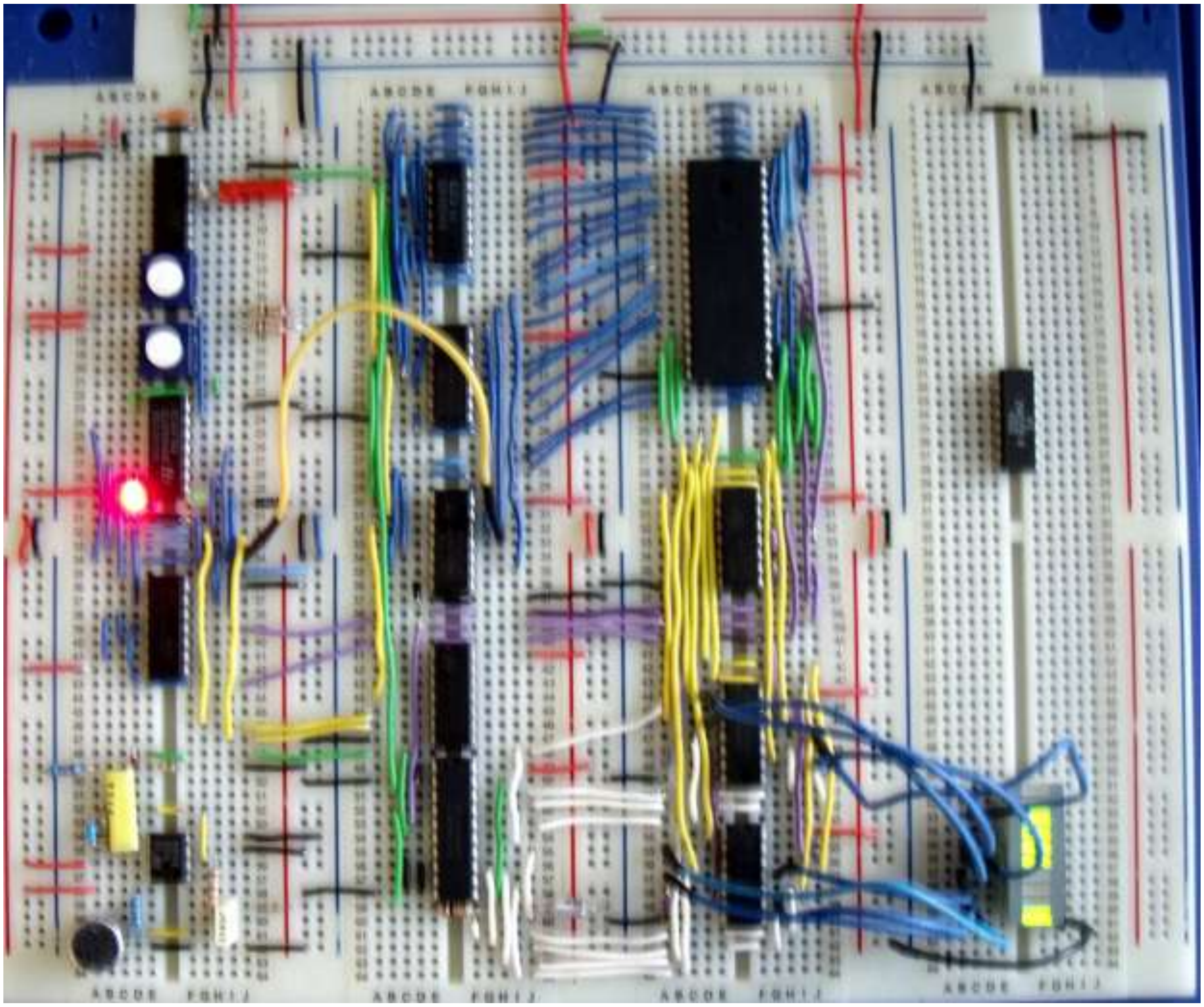
As you can see here, the pulses time nicely with the change in address.

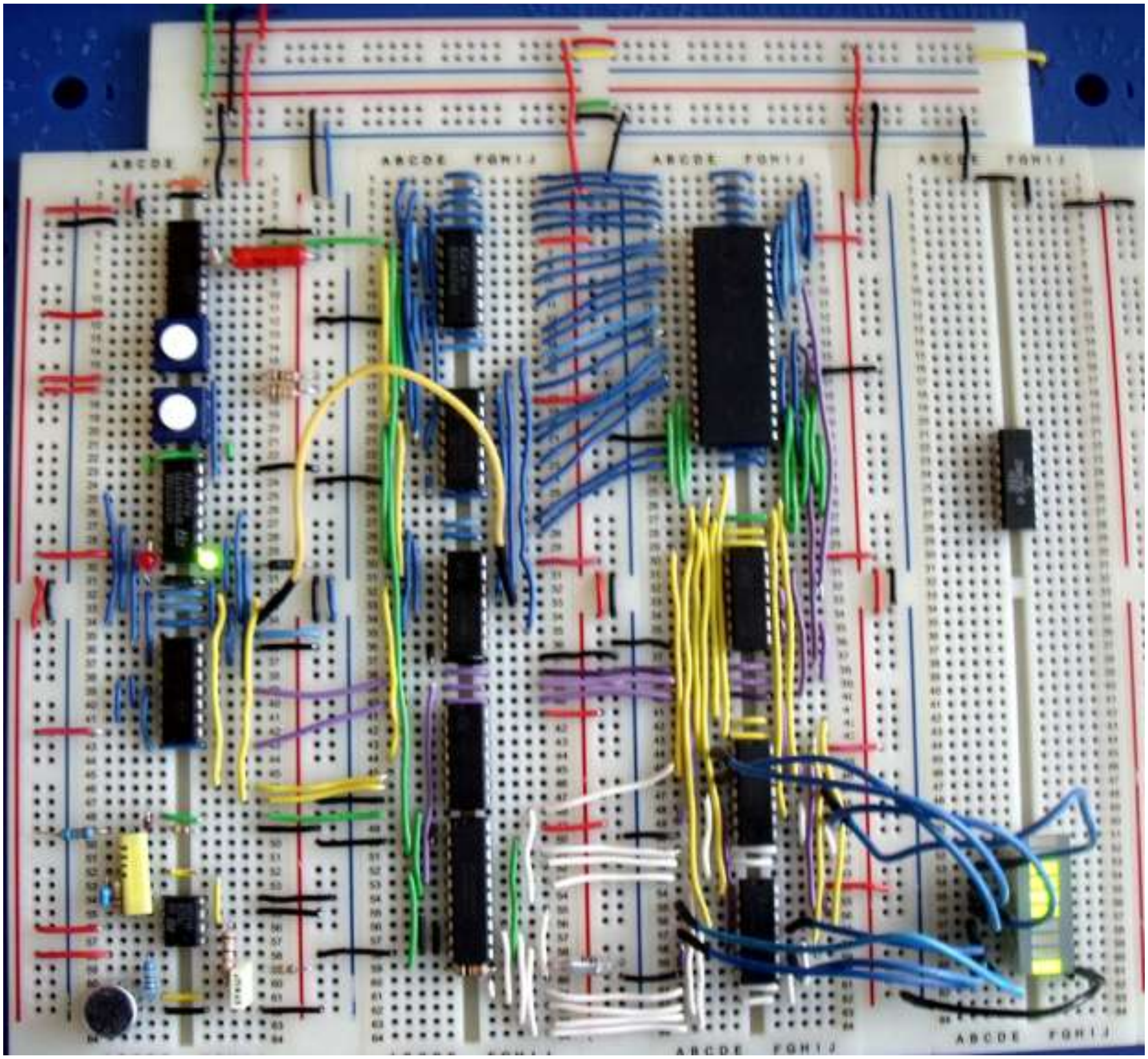


The top line is the WE# line and the bottom the least significant bit. This shows that just as every address is changed, the data from the ADC is stored into it with the WE pulse. This is when the sample is taken.

Subsystem 6 (512Kbit*8bit SRAM Volatile Memory)

Then I connected the ADC to the Analogue switches and the Memory chip, with a bar graph LED display on the line directly from the memory chip.
I would record a pattern by tapping into the microphone and then press the playback button back and see the pattern return on the LED display.

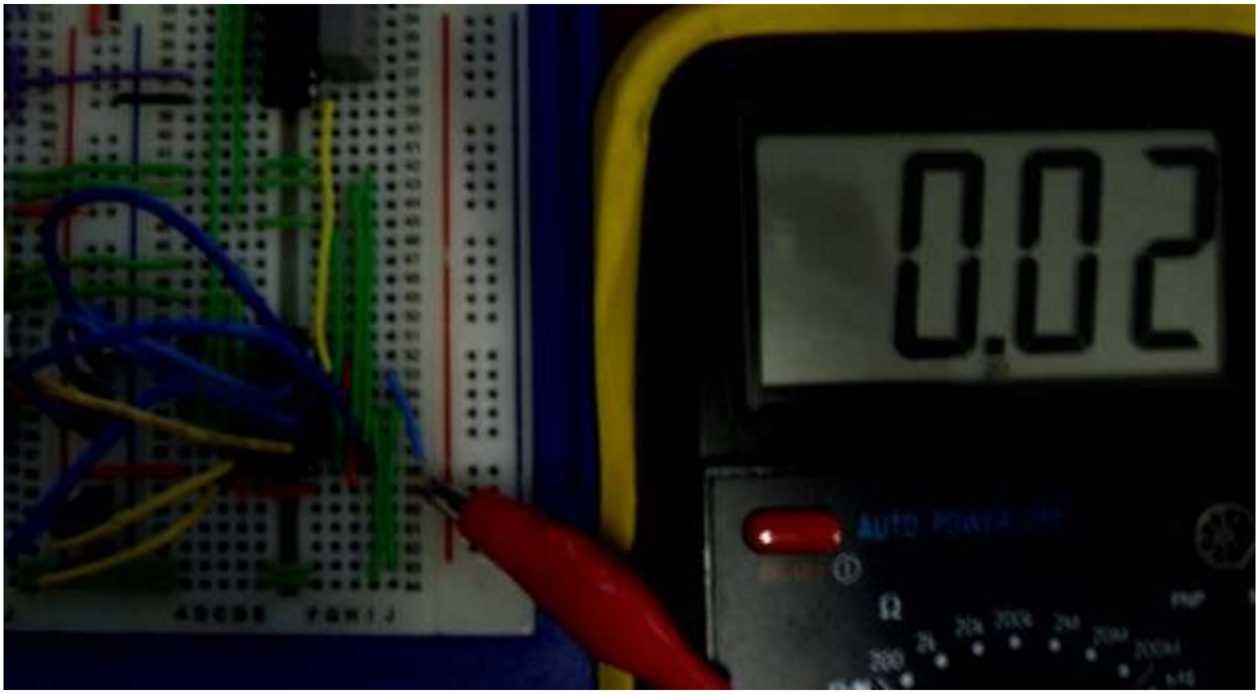




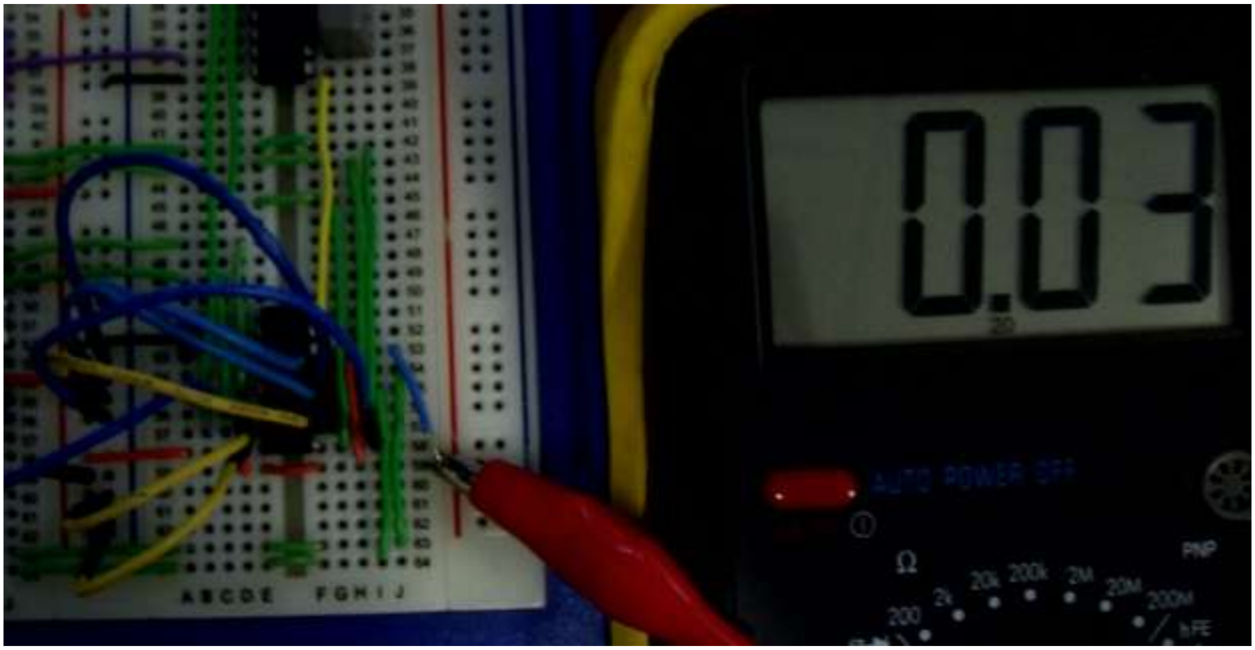
In the first picture the LEDs are lighting with the data going in, but in the second it is coming back from the memory chip. In the same pattern and timing as it went in.

Subsystem 7 (8 Bit Digital --> Analogue converter):

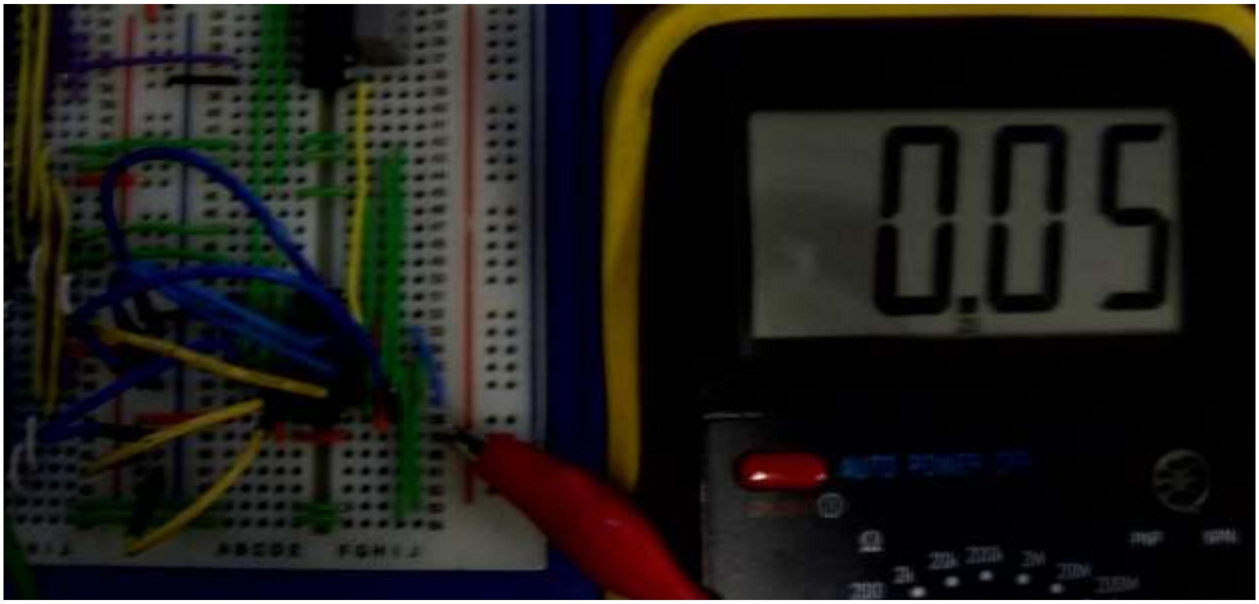
Then I connected up the DAC and started to test it:
When all inputs are low, the DAC outputs 0.02V, which is fine as it will remain stable at this voltage. For this testing I made one line high, and then returned it to low for the next one.



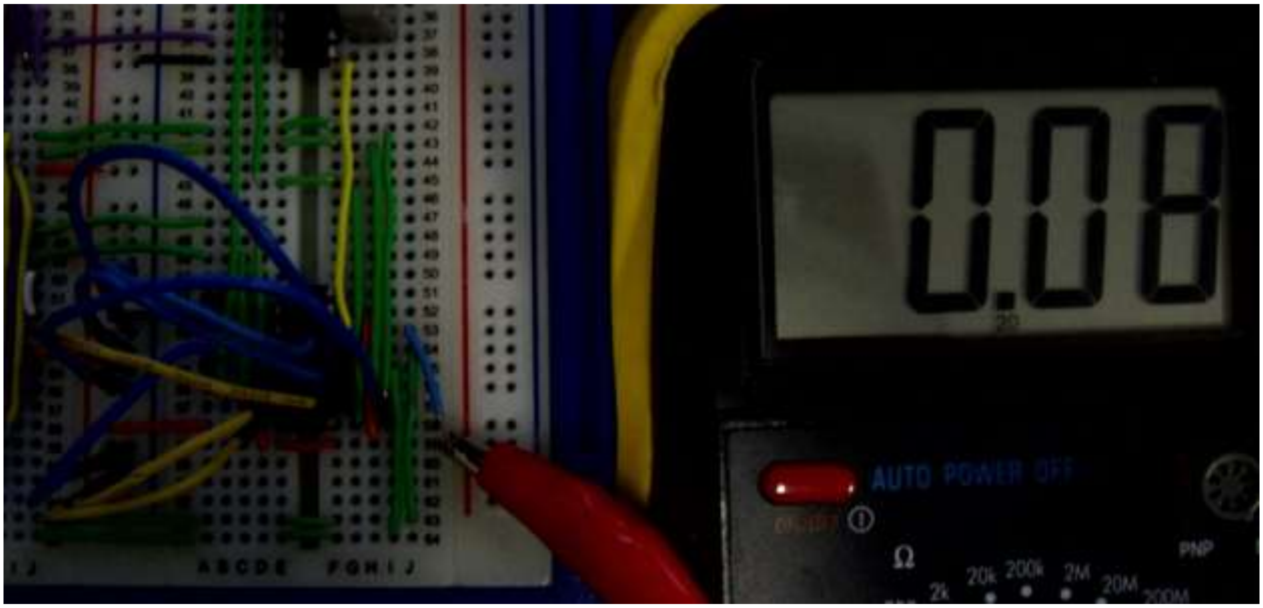
The least significant bit is worth 0.01V, as this adds onto 0.02V when enabled.



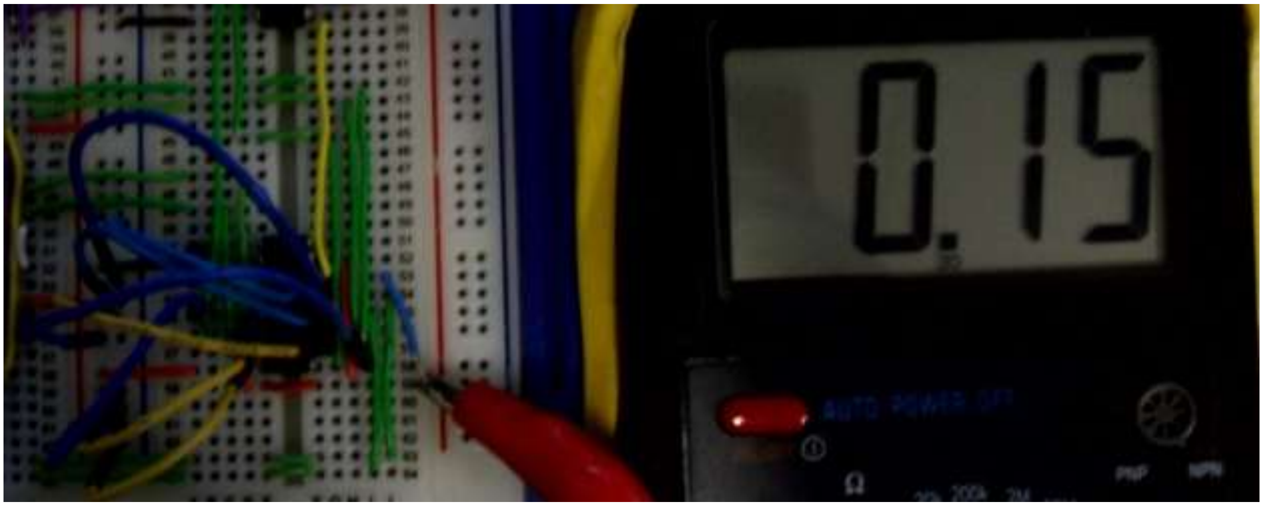
The second bit adds on 0.03V when enabled.



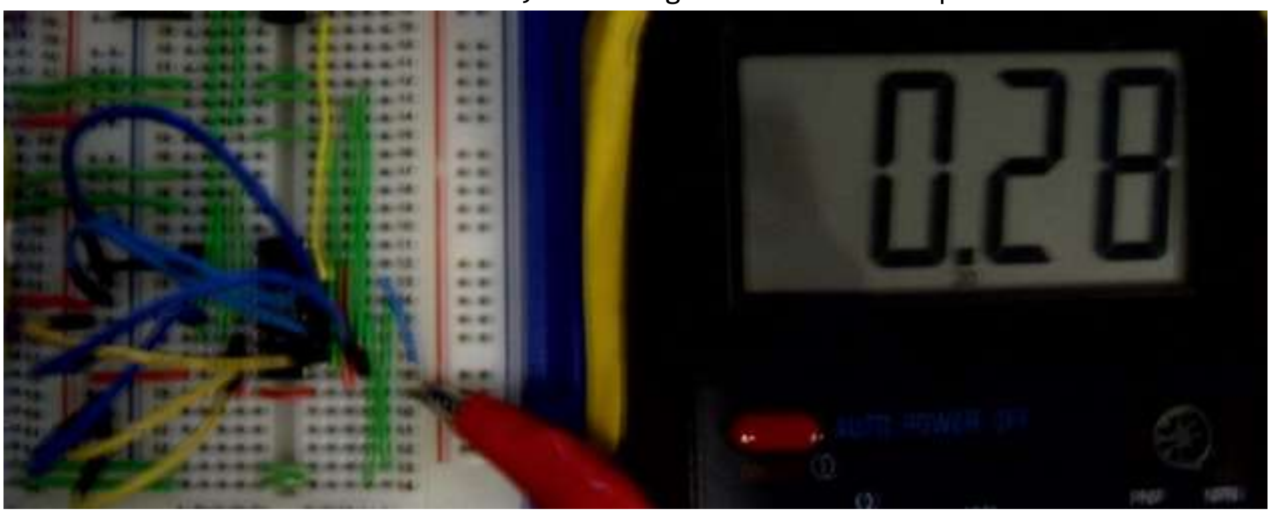
And the third bit adds on 0.03V.



The fourth adds on 0.13V.

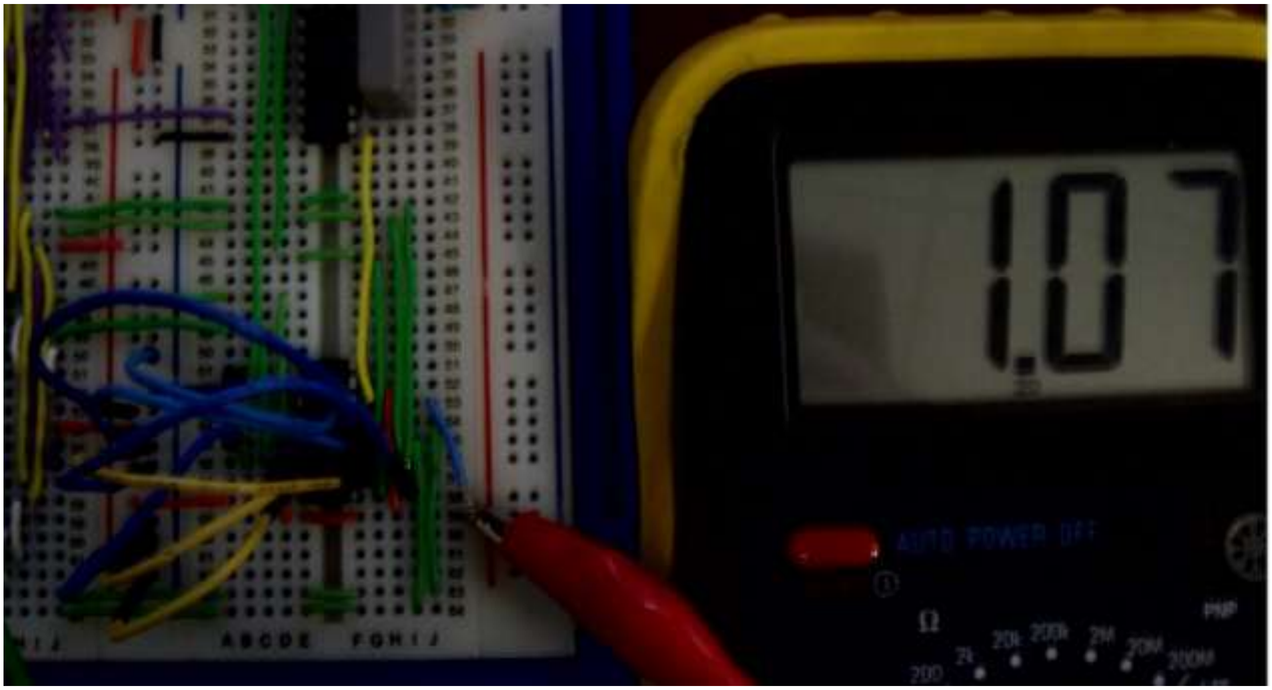


And then the fifth adds on 0.26V, doubling the fifth as expected.

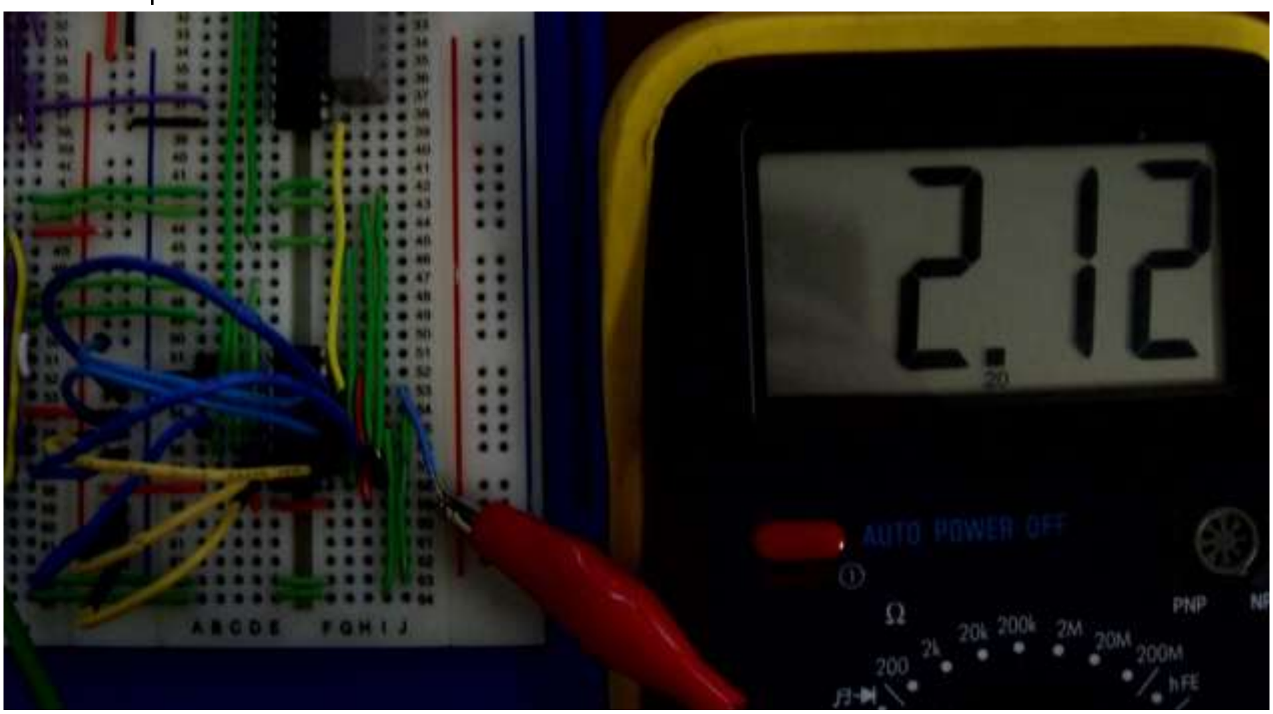


And then the sixth adds on 0.52V, doubling again.

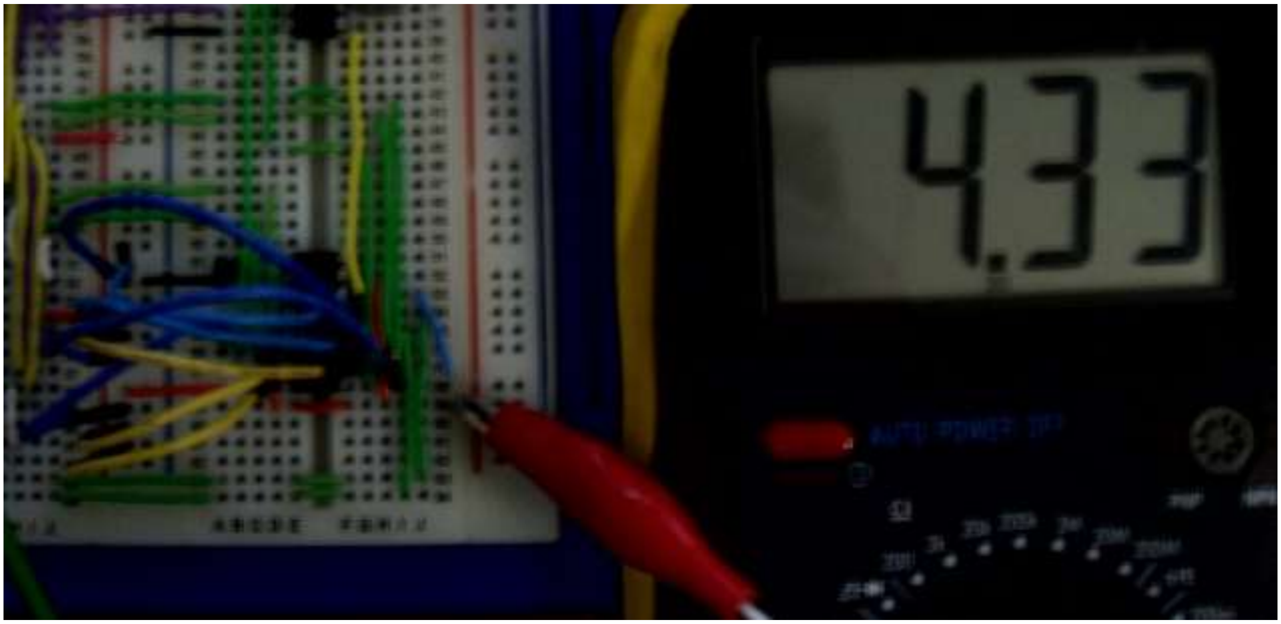
The seventh adds on 1.05V, which is practically doubling again.

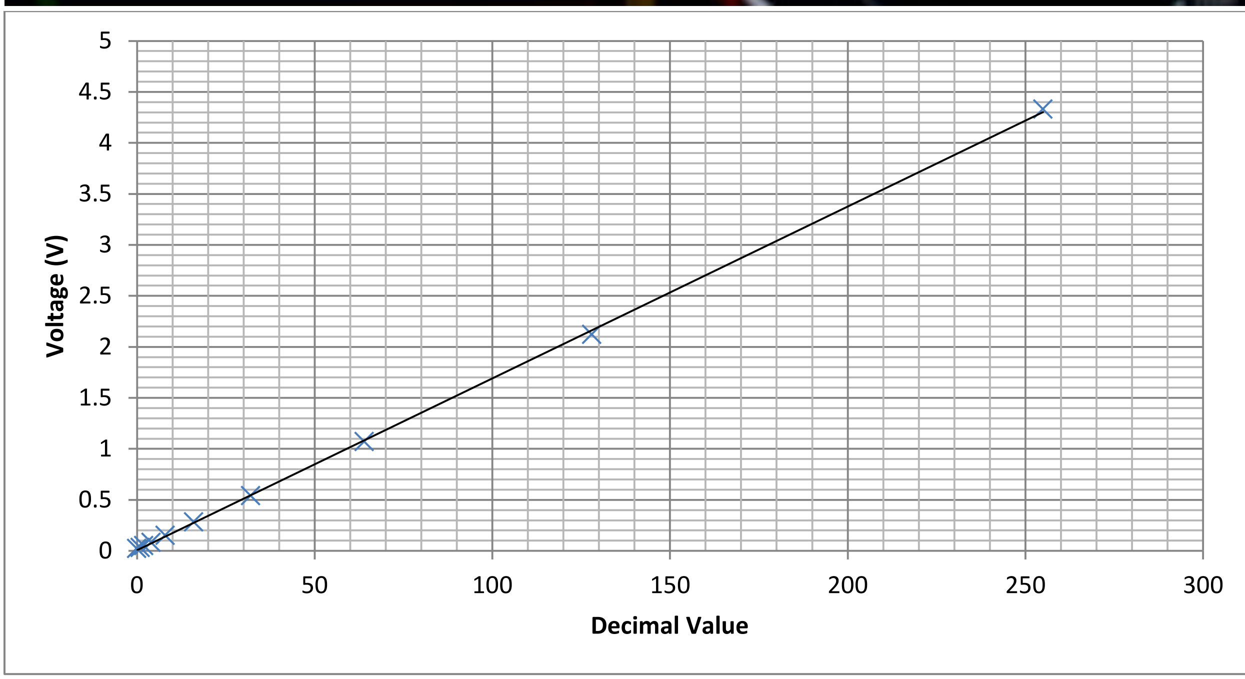


The most significant bit adds on 2.1V which is also doubling, so the results are as expected.



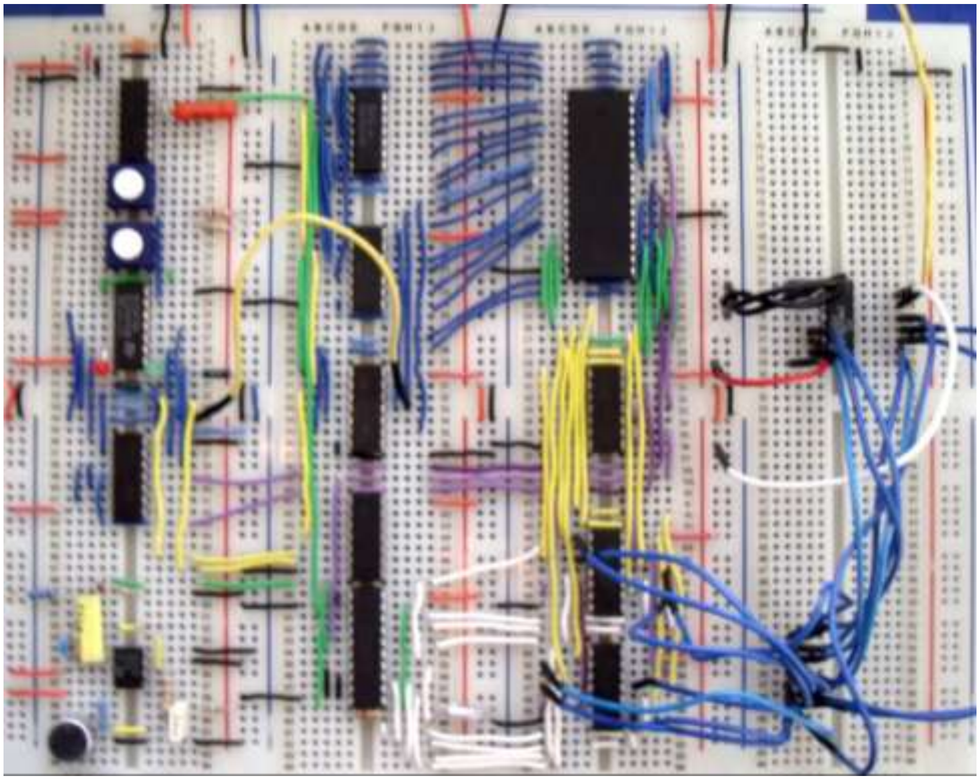
With all the bits high, the DAC outputs 4.33V.

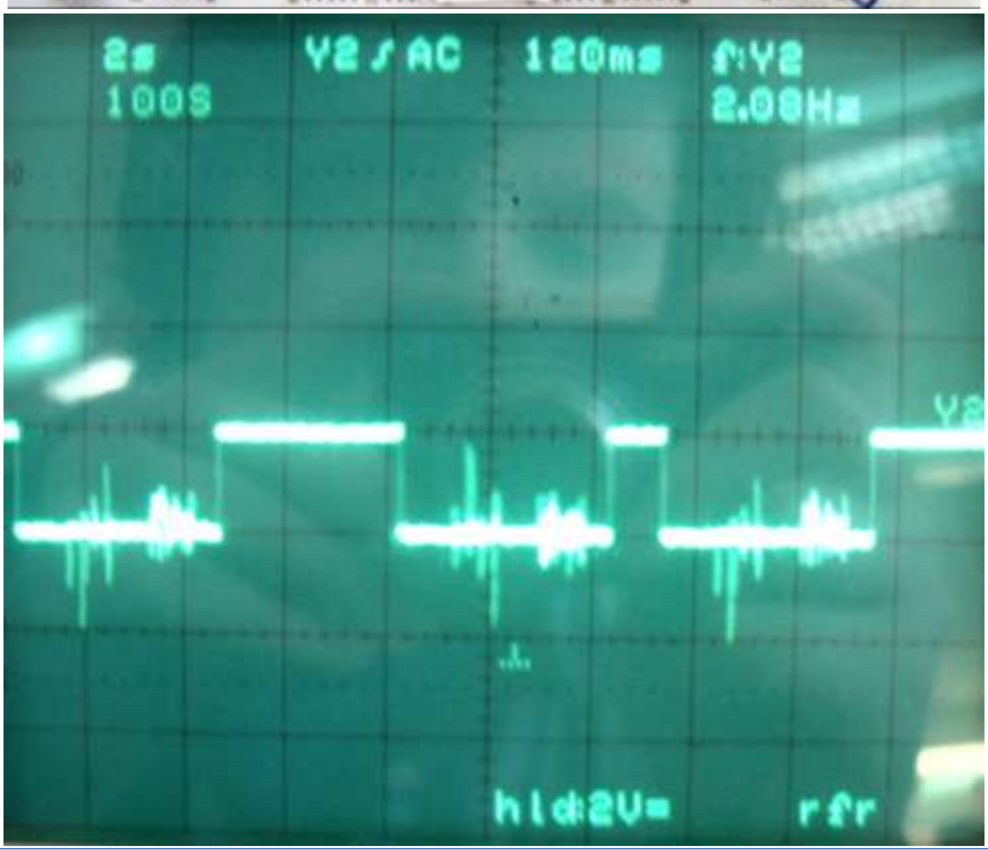




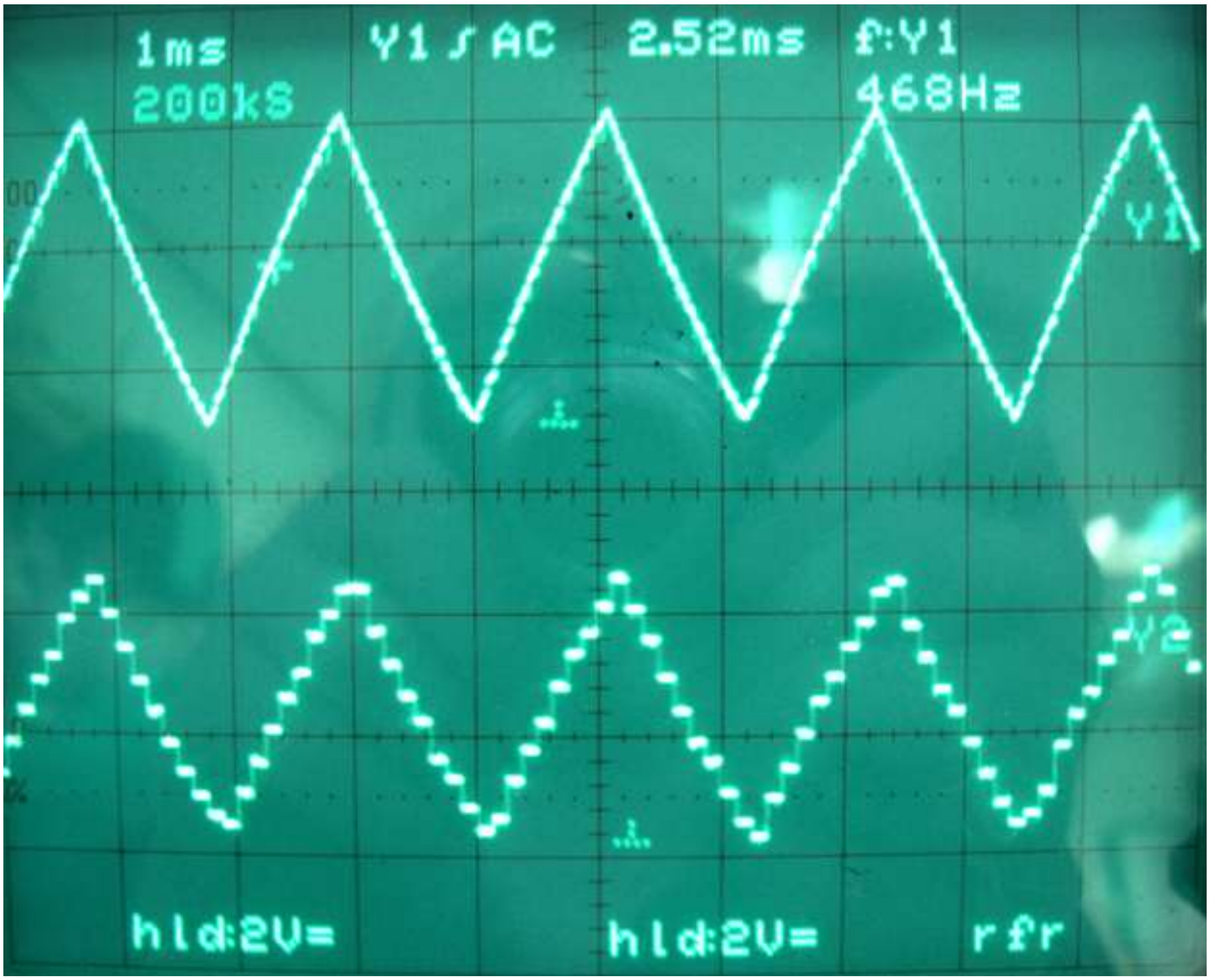
This gives a very nice, smooth linear graph to show that its accurately can reproduce the shape of the input, just with errors in the amplitude, but as this will be altered with the drivers anyway this does not effect the sound.

Then I connected the output of the memory chip into the DAC, and connected the output to an oscilloscope.





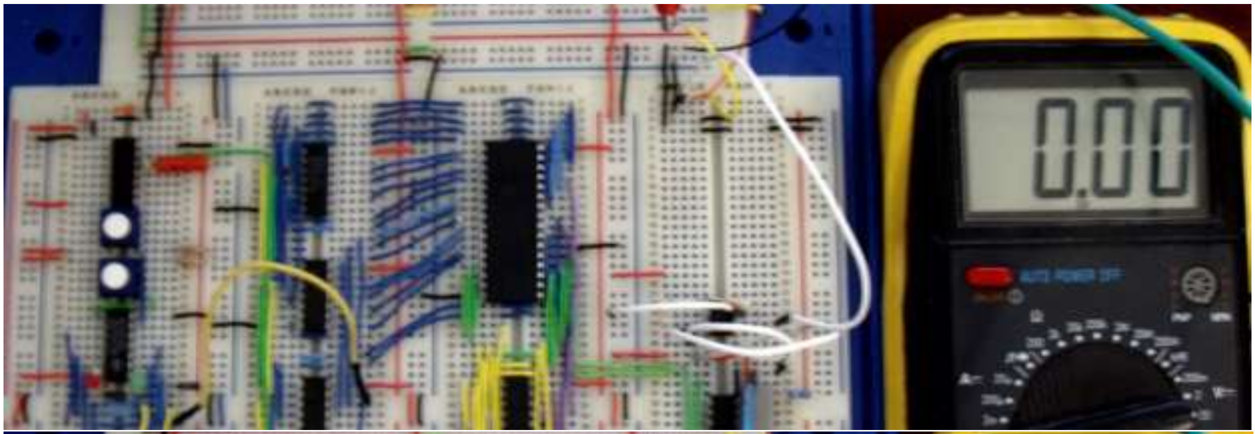
The first section is the raw data from the ADC going back through the DAC. The second two are both playbacks from the memory chip. While the waves do not look perfectly similar, I put this down to the oscilloscope not being able to show the same points exactly as those peaks were very small in time. They were taps on the microphone while the later parts are speech.
At this same time I connected a basic pair of headphones to the output and earth and listened during playback. I could hear what was recorded! However the signal was very quiet as there was no driver for the audio yet and it was still running on what the DAC could drive, which was very little.
There was one major issue at this stage that could not be fixed however. This was that in between the samples, instead of sloping up to the next sample, the DAC just stepped up.
This can be seen here with this sine wave on the input and output of the ADC --> DAC circuit.

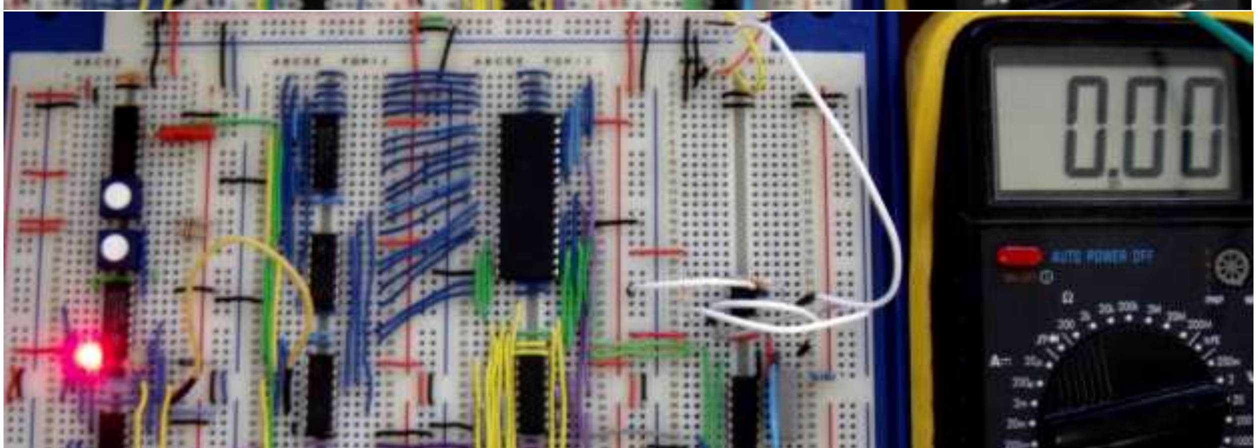


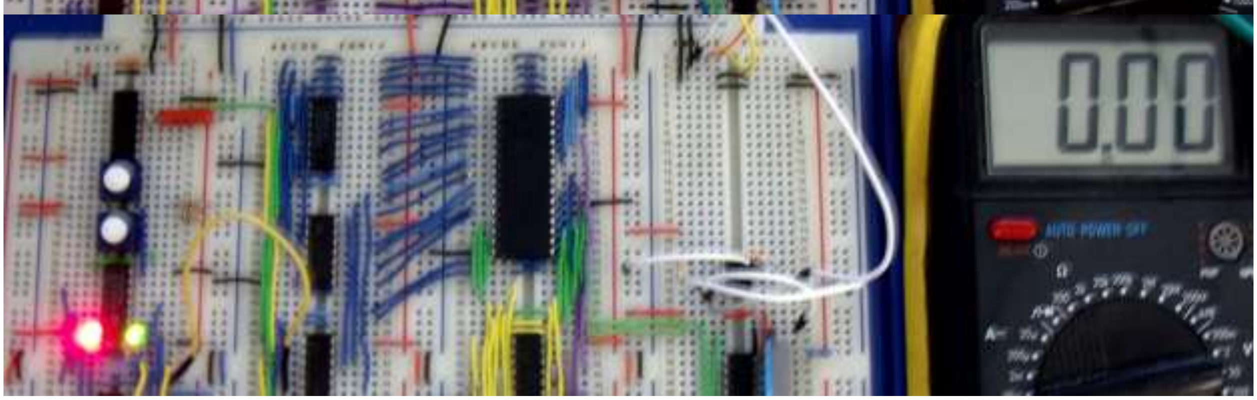
This issue, I believe to cause some definite bad quality from the recording. And inputting a circuit to smoothen this out would cause too much complexity and take up more space which is very limited with the scale of my project.

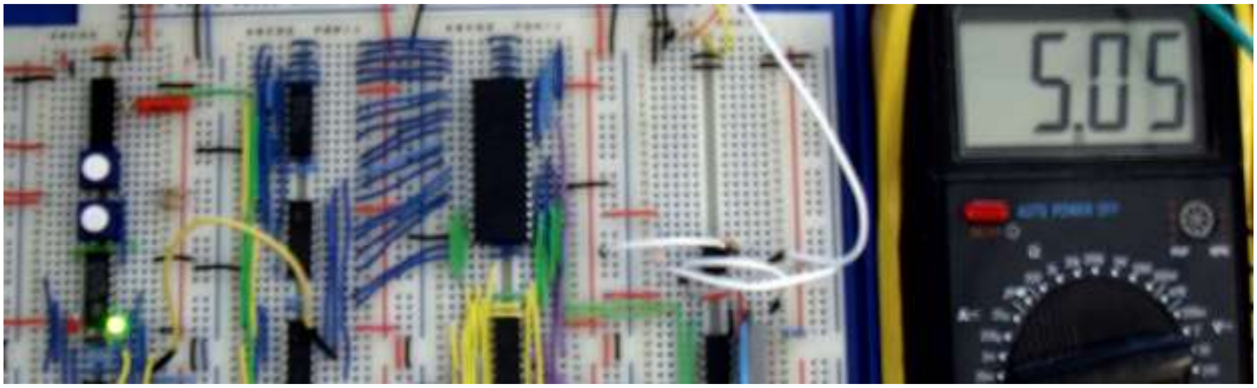
Output Enable:

Then I connected up the Logic gates and analogue switch that would go between the DAC and the driver. As you can see it works perfectly and the Analogue switch is only enabled when the playback button is on.







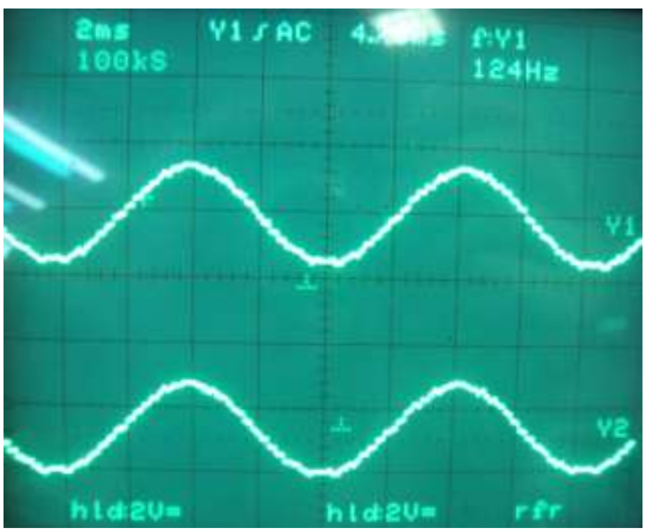


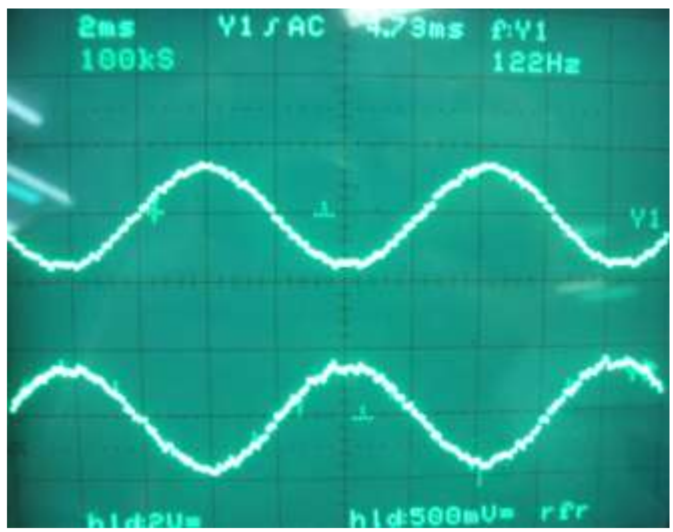
Subsystem 8 (Driver and Loudspeaker):

Voltage offset and primary amplifier:

The first picture is where the signal has emerged out of the DAC and gone through the RC circuit to bring the signal back to Positive and negative around 0v. This does not seem to change the signal at all apart from bring the voltage offset down at an important frequency range.

On the second amplifier the signal is reduced by the correct amount. Now being just under 500mV in amplitude, but there are a few random peaks appearing during this. This should not be audible but it is visible and could

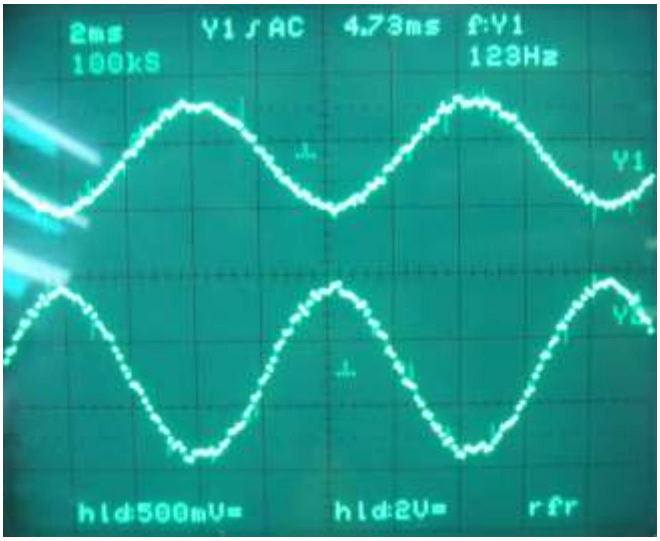


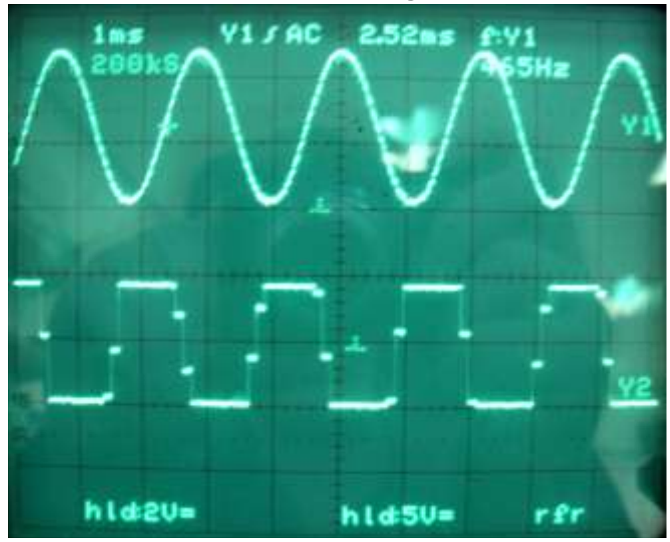


pose a problem if complete audio quality was wished for.

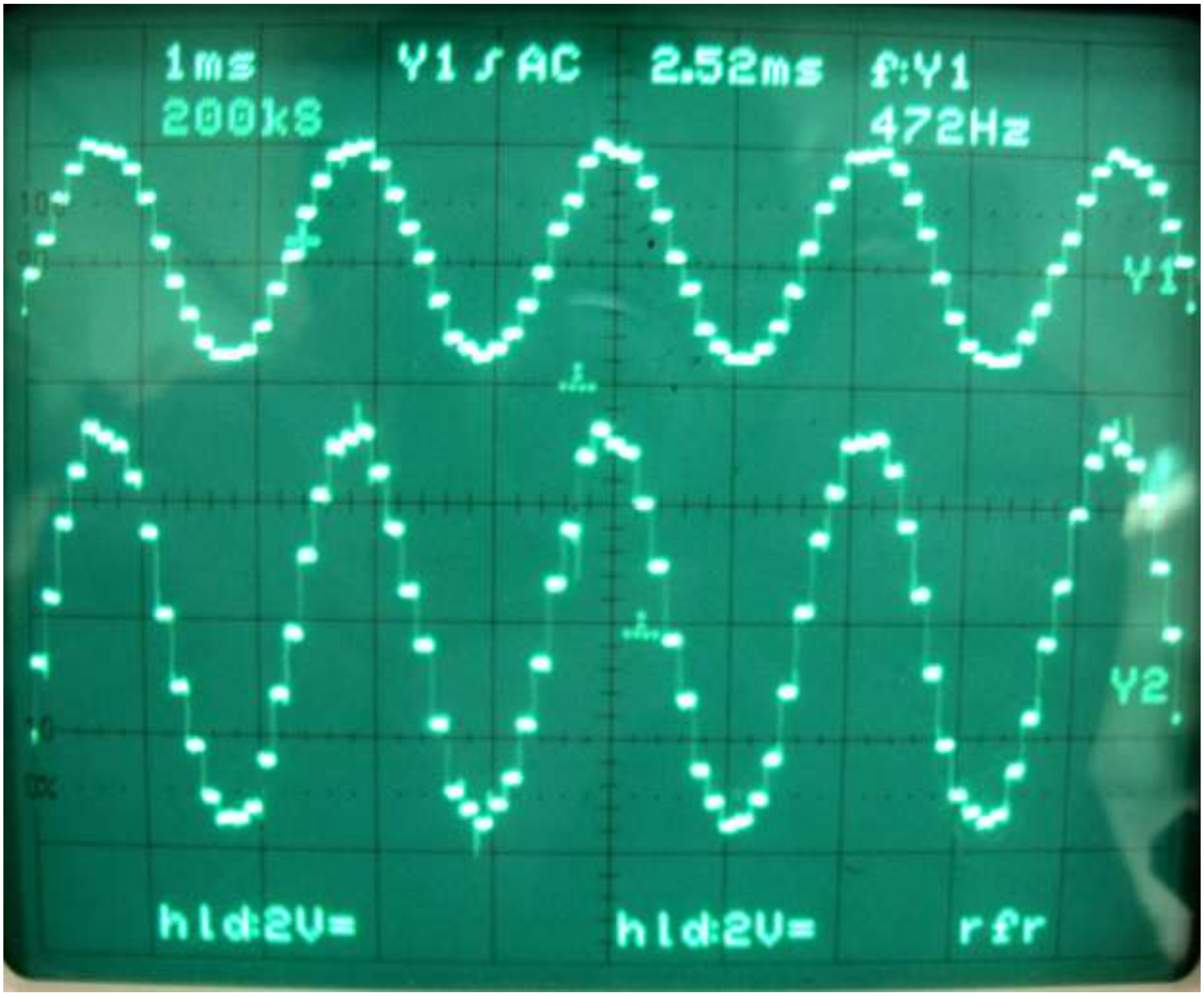
Second amplifier and Loudspeaker:

The photo on the left is the signal from the primary amplifier and then out of the audio amplifier. The gain is just above 4. While on the right we have





an extremely high gain. Where the signal saturates (the top line on the right photo is from the sine wave input of the entire circuit, not the final amplifier)



This is another picture of the signal from the Output of the DAC to the output of the final amplifier. While there are the stepping issues caused by the DAC, but the output signal looks almost identical but with higher amplitude and the random blips of noise brought on by the primary amplifier. While unable to be noticed on an oscilloscope, the bottom line can source a far larger current too. It is safe to say that between the DAC output and the loud speaker not much noise is added on, apart from random peaks by the 741.

Complete System Testing, Analysis and Conclusion

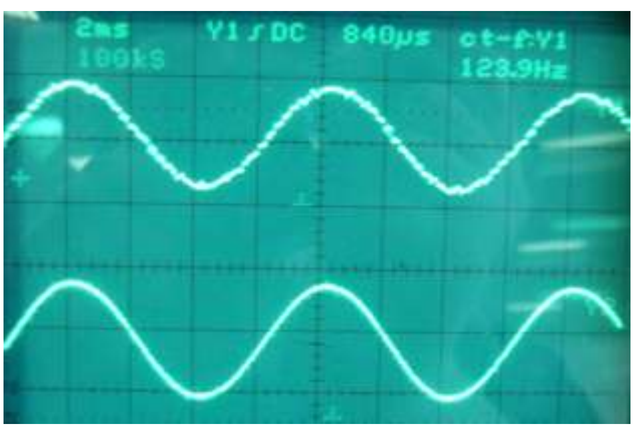
Issues found and their causes:

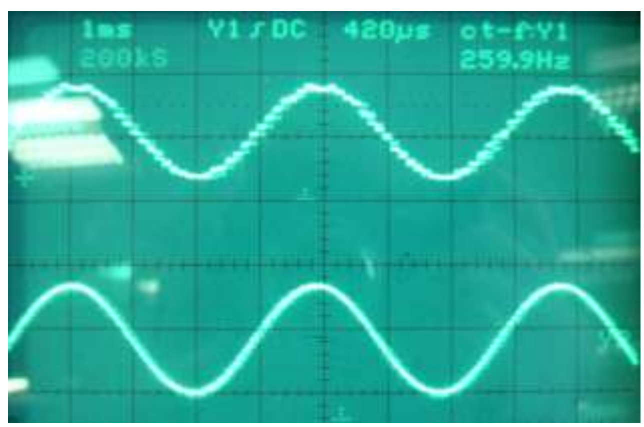
The first issue that I discovered while testing the entire circuit was that sometimes when turned on one of the latches would trigger automatically and start its cycle. Luckily it would self-correct itself by just doing a counter depending what the counter was set to be. It would always be both LEDs so no sound would come out of the speaker during this.

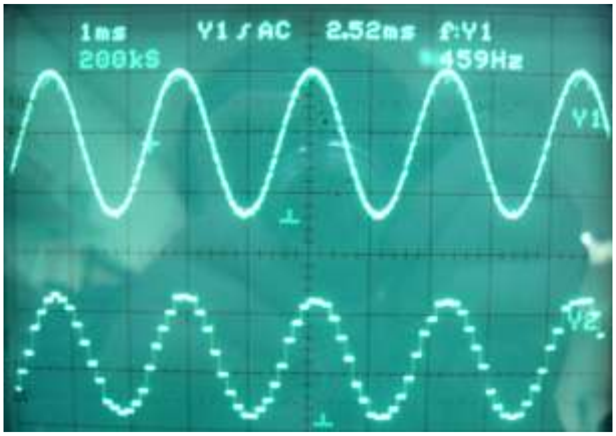
Another issue that I discovered was that during recording, a small amount of what was being recorded would come out of the loud speaker. I put this down to noise caused by the system while recording which isn't fully blocked out by the capacitors and analogue switch.

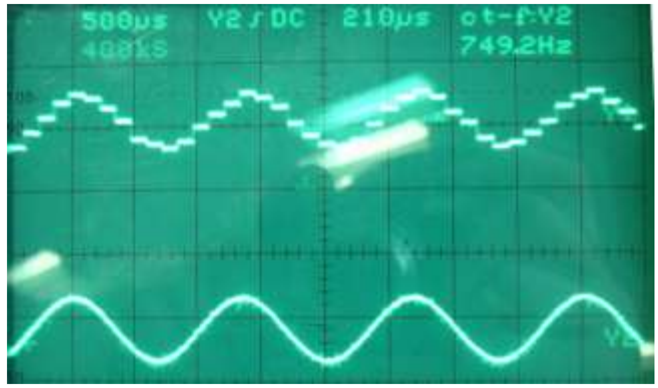
The third issue that I discovered was that the audio was not very clear and had a lot of noise. This issue was hard to track down and still could be multiple possibilities. First, the microphone is not of the best quality and the quality of the sound can be increased greatly by making sure that the sound you want to record is at the correct levels otherwise it will either saturate at the preamp or be overruled by noise.

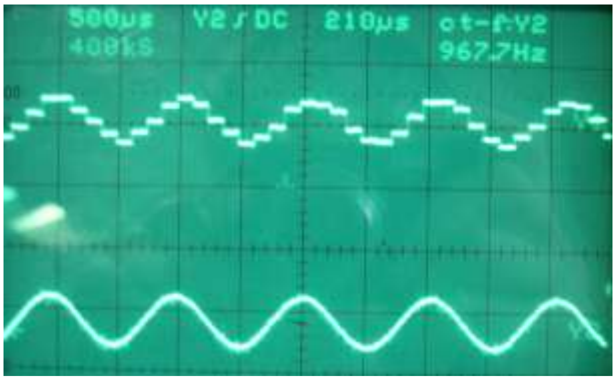
Another possible cause for this would be that the DAC steps the voltage up, instead of creating a smooth curve from the samples, just steps voltage up or down on each sample, this can be seen in the following photographs. In the first photograph, at a low frequency like 120Hz, this stepping has very little effect. However, as it gets higher and higher frequency, you can see it slowly makes more and more effect, and then as it gets to near 1KHz and it is majorly starting to change it. Any higher and the wave is almost unrecognisable. While I calculated that 4000Hz should be used to stop aliasing from the sample rate, this problem causes issues as well that were not expected. However this is good enough quality to still hear voice clearly and be able to recognise what song has been stored into it without fixing such issue.

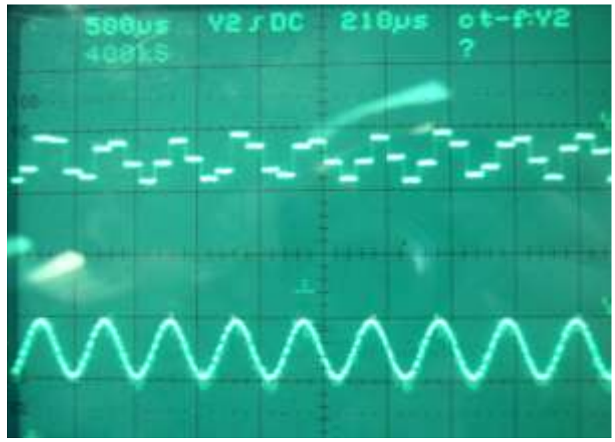






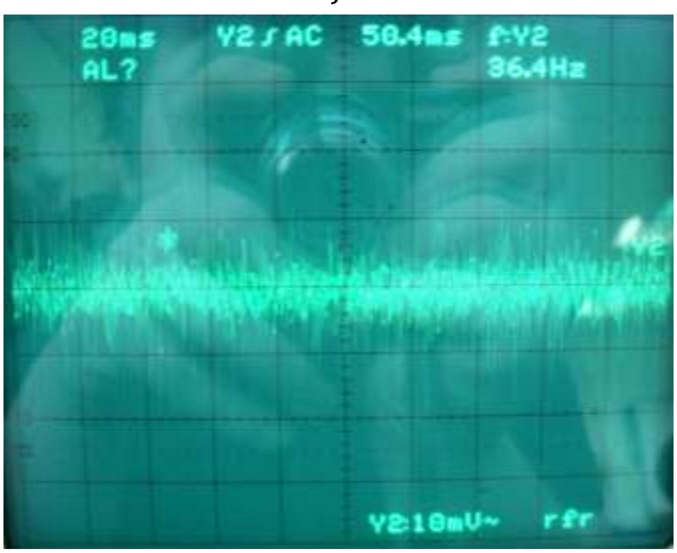






In case the noise was being created by general noise from the power supply, I connected up an oscilloscope onto the 5v rail via the AC coupling so I could see the noise of it.

From the looks of it, while there is some noise on the line, this would not be enough for a major noise to be heard as much as it is. Mostly around 5mV of change, with some peaks hitting 30-40mV



Another issue that I found was that when you turn it on, the memory chip has random data stored in it which has no correspondence to before it turned off, and if you press the replay button then you hear a terrible amount of white noise.

What I would do to fix these issues if I had the resources:

A quick fix to the stepping issue would be having an ADC with a much faster conversion rate. This would allow me to increase the sample rate dramatically so I could get many samples that the stepping would be near impossible to see in audible frequencies, this would take a very fast ADC though, which can cost a lot of money and would take any storage equally fast.

Or instead of this, building a subsystem that would connect up the samples with a straight line or other suitable voltage increase/decrease between these values. I believe this could increase the sound quality a lot, especially when the frequencies are greater than 500Hz

Another upgrade that would be good would be non-volatile memory instead of volatile which would allow for data to be stored even when the device has been turned off, and when turned on it would not be full of noise.

Overall conclusion of the project:

My project will turn on, record audio and play it back at a decent enough quality for words to be made out, and music replayed at a fairly enjoyable level. While there are some definite issues with the system, these do not majorly effect the way it works or cause issues that cannot be ignored. If a raw sine wave is put in, then the wave will seem very accurate to the same wave on the other side, minus the stepping at higher frequencies.

The noise while could be annoying to some is a minimal if recording conditions are optimum and for use of voice recording will not be an issue. If you use a media player to sound loud music next to the microphone it comes out a lot better quality than when you speak into it from a distance.

If the data is injected direly in from a headphone connection on a media player, ignoring the microphone and preamp the audio quality for the lower frequencies does increase. However, due to it missing the preamp this does not get filtered, so perhaps in a future system if both input types were allowed then the filter and preamp would have to be on a different amplification system.

The lack of Non-volatile memory can be annoying but if this was made into a device there are low power modes which some of the chips can be put into and the only chip that requires power would be the memory chip; so while it being volatile is not the best, it is easily good enough with easy solutions to potential problems.

Compared to modern electronics, my circuit is huge. Modern electronics consisting just of a single microprocessor and possibly some flash memory could do what this circuit does and far more. But those require programming and are far harder to troubleshoot if something goes wrong. This also has a far simplified control system.

For the future I would like to add a device which would allow me to pause recording and then continue from the same spot. Perhaps that would be a clock enable line. Or a method of stopping recording and then recording a second two second recording and being able to choose between them.

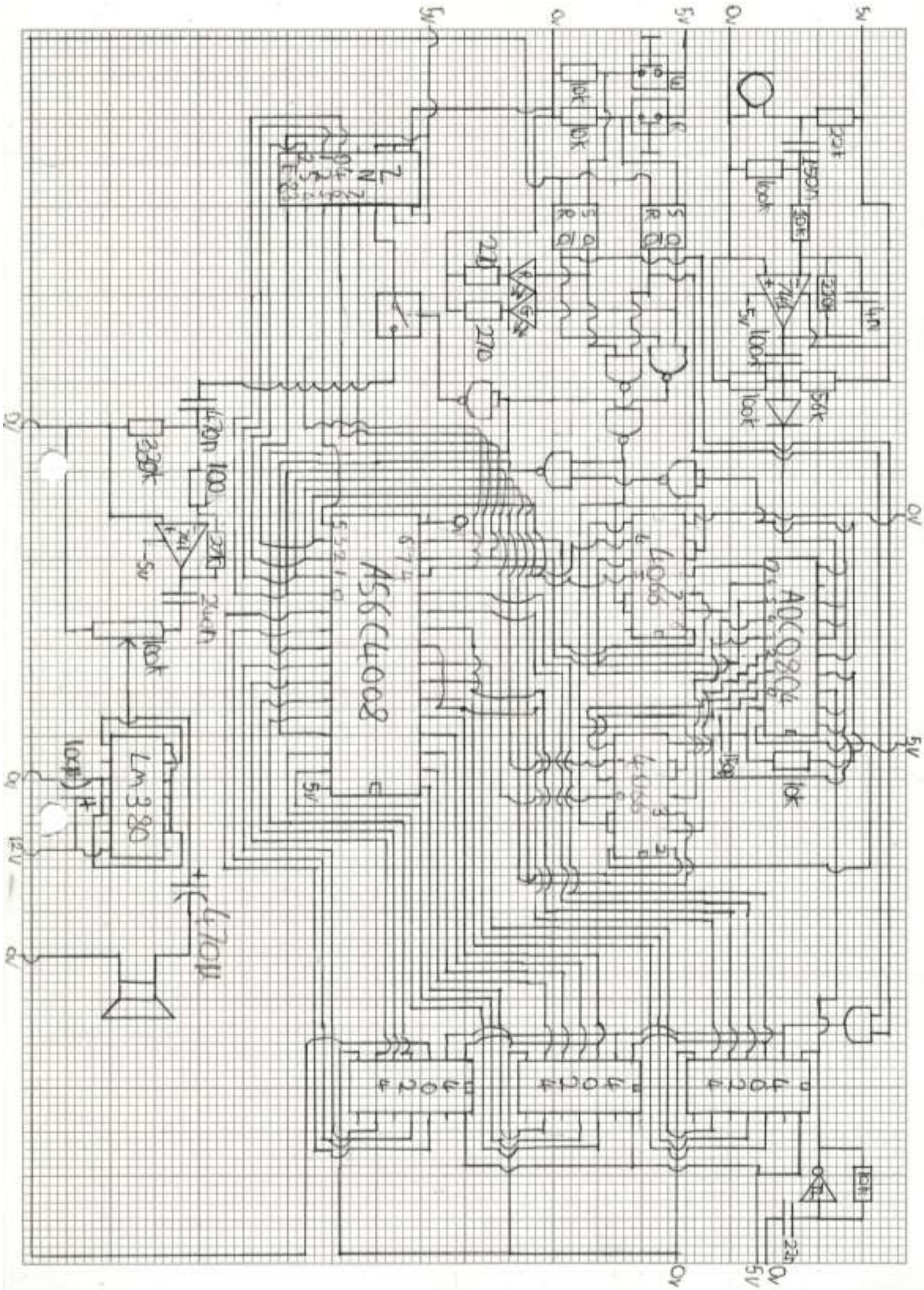
Another thing that I would like to change is the current method for timing settings, a jump lead is a very rugged method for switching between these. Perhaps in the future I would have a dial which could pick between them, or a 3 bit switch digitally connected to a multiplexer to choose between the settings.

While this system is built for recording audio, it can be used as a data logger and replay system for any device. As long as you can match up the input values correctly, so that the input voltage was the correct amount for the voltage offset or ADC. Then all you need is a little modification to replace the microphone. Then you could use this to data log temperature, or anything else that you had an electric probe for. You could also slow down the clock if you wanted to record this over a long time, and then speed it up to replay it, perhaps with a potentiometer in the relaxation oscillator to change the frequency of the clock.

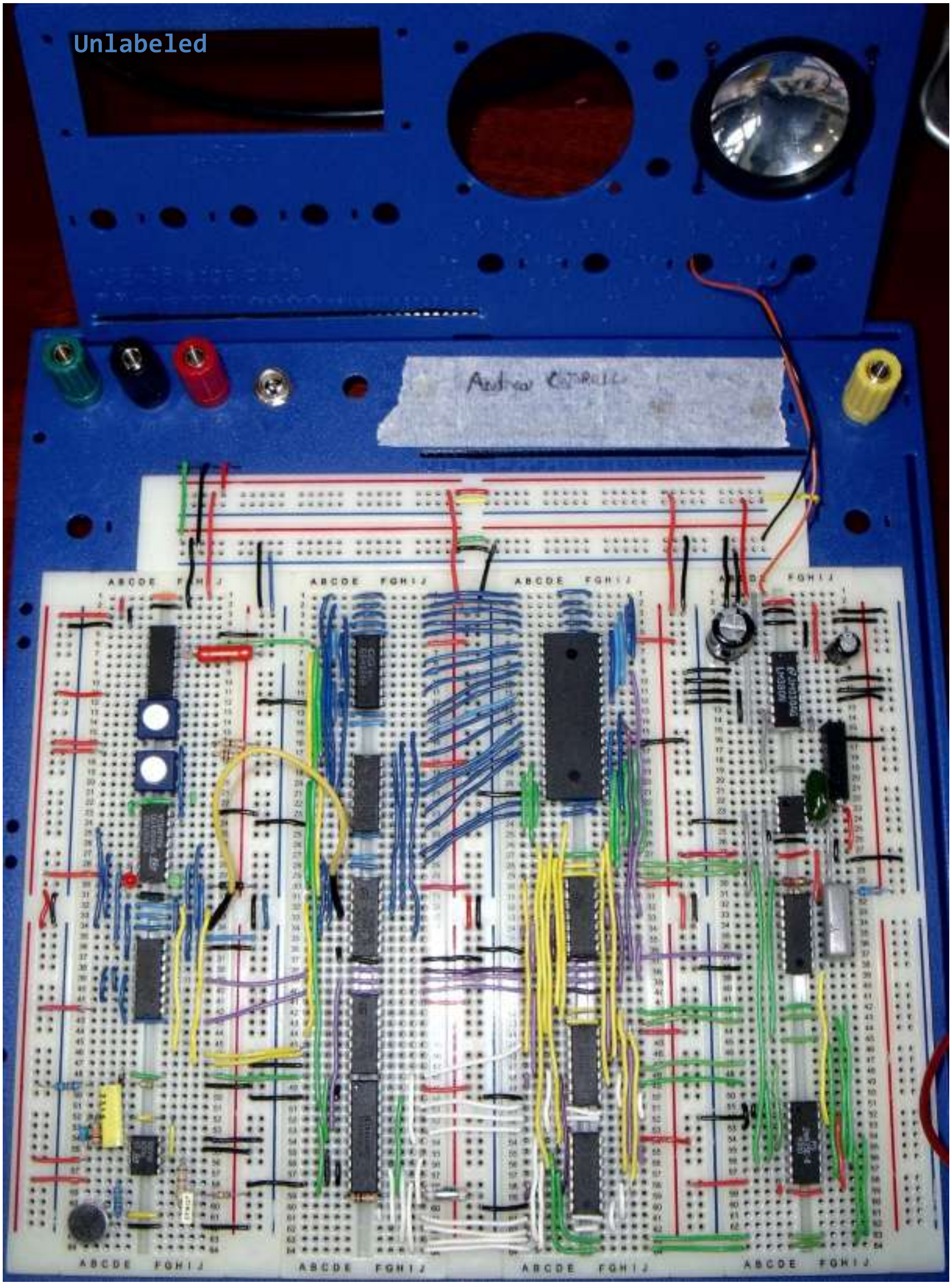
Another thing in the future that I would like to do is to somehow create a stereo recorder. By adding a second microphone and speaker, and having a multiplexer switch between each as it records and plays back. However for this I would need to either double the frequency or put up with half as good quality. But if I chose to double the frequency I would have to obtain a better ADC which could go faster than the current on. This would also then allow me to reach higher frequencies and better quality sound. Perhaps also then you could have an ADC that could record positive and negative values from one signal line as well which would remove the need for the initial voltage offset and make this system simpler.

Finally; I would like to somehow make it run off a power supply so that I only need 3 voltages, instead of 4. Or perhaps even 2 as having a power supply unit that would supply all 4 voltages can be difficult to obtain and can also be expensive, which if for a general consumer would be irritating.

Circuit Diagram



Photos of Entire Circuit



Labeled

5V

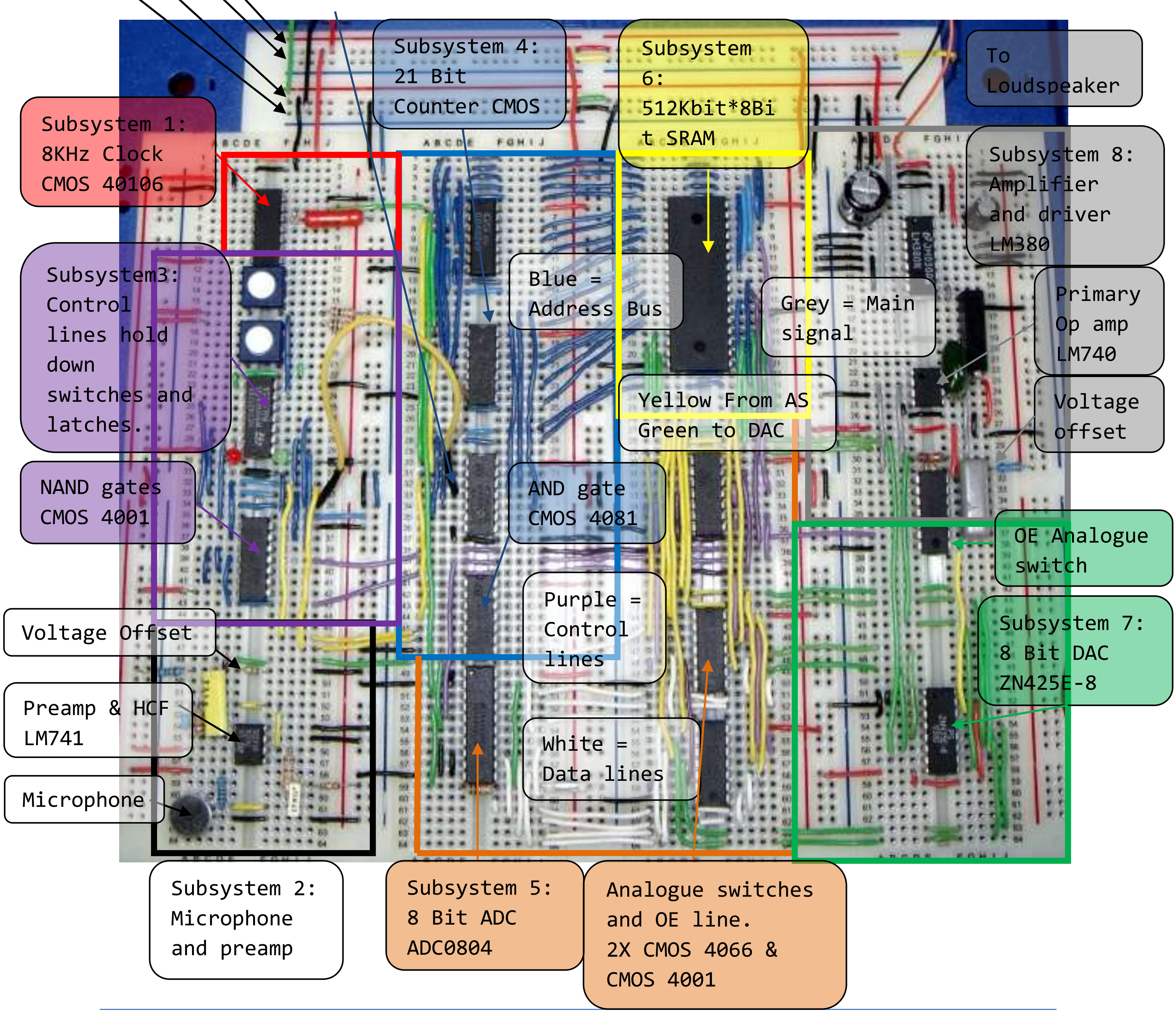
12V

0V

-5V

Jump lead sets time:
Row 32, Right = 2 Seconds
Row 33, Right = 4 Seconds
Row 35, Right = 8 Seconds
Row 35, Left = 16 Seconds
Row 34, Left = 32 Seconds
Row 33, Left = 64 Seconds

Subsystem	Colour	Subsystem	Colour
1	Red	5	Orange
2	Black	6	Yellow
3	Purple	7	Green
4	Blue	8	Grey



Full component list

Category	Specification	Quantity
Wire	Red	0.896M
	Black	1.222M
	Purple	0.929M
	Yellow	1.199M
	Blue	2.363M
	Grey	0.23M
	Green	1.76M
	White	0.802M
	Jump	1
Resistors	270Ω	2
	10E+3 Ω	5
	22E+3 Ω	1
	27E+3 Ω	1
	56E+3 Ω	1
	100E+3 Ω	3
	220E+3 Ω	2
Potentiometer	100E+3 Ω	1
Capacitors	240E-9F	1
	150E-9F	1
	100E-9F	1
	47E-9F	1
	22E-9F	1
	4E-9F	1
	150E-12F	1
Electrolytic	470E-6F	1
	100E-6F	1
CMOS 4000 Chips	40106	1
	4011	2
	4013	1
	4024	3
	4066	3
	4081	1
Other Chips	LM741	2
	LM380	1
	ADC0804	1
	ZN425E-8	1
	AS6C4008	1
Diodes	0.7V Silicon	1
	Green Light	1
	Red Light	1
Other components	Microphone	1
	Loudspeaker	1
	Push to make Switches	2
	Power supply	12V,5V,0V,-5V

Bibliography

[1] The useful circuit's booklet – QMC publication

For the microphone circuit: Page 15. Diagram 1

For the driver circuit: Page 25. Diagram 2.

[2] National Semiconductors ADC0804 Data sheet
(http://www.national.com/ds/DC/ADC0801.pdf)